HST-3000

VoIP Testing

User's Guide

JDSU

HST-3000

VoIP Testing

User's Guide

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Ordering information This guide is a product of JDSU's Technical Information Development Department, issued as part of the HST-3000. The catalog number for a printed guide is ML-059701. The catalog number for a USB stick containing all HST-3000 user documentation is ML-060301.

Terms and Conditions

The provision of hardware, services and/or software are subject to JDSU's standard terms and conditions available at www.jdsu.com.

Federal Communications Commission (FCC) Notice

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) This device must accept any interference received, including interference that may cause undesired operation.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

– Reorient or relocate the receiving antenna.
– Increase the separation between the equipment and receiver.
– Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
– Consult the dealer or an experienced radio/TV technician for help.

In order to maintain compliance with the limits of a Class B digital device JDSU requires that quality interface cables be used when connecting to this equipment. Any changes or modifications not expressly approved by JDSU could void the user's authority to operate the equipment.

Industry Canada Requirements

This Class B digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe B est conforme à la norme NMB-003 du Canada.

WEEE and Battery Directive Compliance

JDSU has established processes in compliance with the Waste Electrical and Electronic Equipment (WEEE) Directive, 2002/96/EC, and the Battery Directive, 2006/66/EC.

This product, and the batteries used to power the product, should not be disposed of as unsorted municipal waste and should be collected separately and disposed of according to your national regulations. In the European Union, all equipment and batteries purchased from JDSU after 2005-08-13 can be returned for disposal at the end of its useful life. JDSU will ensure that all waste equipment and batteries returned are reused, recycled, or disposed of in an environmentally friendly manner, and in compliance with all applicable national and international waste legislation.

It is the responsibility of the equipment owner to return equipment and batteries to JDSU for appropriate disposal. If the equipment or battery was imported by a reseller whose name or logo is marked on the equipment or battery, then the owner should return the equipment or battery directly to the reseller.

Instructions for returning waste equipment and batteries to JDSU can be found in the Environmental section of JDSU's web site at www.jdsu.com. If you have questions concerning disposal of your equipment or batteries, contact JDSU's WEEE Program Management team at WEEE.EMEA@jdsu.com.

Contents

About This Guide

This chapter describes how to use this guide. Topics discussed in this chapter include the following:

Purpose and scope

The purpose of this guide is to help you successfully use the features and capabilities of the Acterna HST-3000.

This guide includes task-based instructions that describe how to configure, use, and troubleshoot the HST-3000's VoIP testing option.

Assumptions

This guide is intended for novice, intermediate, and experienced users who want to use the HST-3000 effectively and efficiently. We are assuming that you have basic computer experience and are familiar with basic telecommunication concepts, terminology, and safety.

Terminology

The following terms have a specific meaning when they are used in this guide:

– **HST-3000** — Handheld Services Tester 3000. In this user's guide, "HST-3000" is used to refer to the HST-3000 family of products or to the combination of a base unit and attached SIM. "HST" is also sometimes used to refer to the base unit/SIM combination.
– **SIM** — Service Interface Module. Sometimes referred to generically as the module. The SIM provides test application functionality.

For definitions of other terms used in this guide, see "Glossary" on page 61.

Application-oriented user guide

The *HST-3000 VoIP Testing User's Guide* is an application-oriented user's guide contains information relating to the basic use of the device and includes detailed procedures for testing and troubleshooting VoIP service.

This user's guide should be used in conjunction with the *HST-3000 Base Unit User's Guide*.

HST-3000 base unit user's guide

The *HST-3000 Base Unit User's Guide* contains overall information relating to device and general functions such as using the unit with a keyboard, peripheral support, battery charging, printing results, and managing files. This guide also contains technical specifications for the base unit and a description of JDSU's warranty, services, and repair information, including terms and conditions of the licensing agreement.

Safety and compliance information

Safety and compliance information are contained in a separate guide and are provided in printed format with the product.

Technical assistance

If you need assistance or have questions related to the use of this product, use the information in Table 1 to contact JDSU's Technical Assistance Center (TAC) for customer support.

Before you contact JDSU for technical assistance, please have the serial numbers for the service interface module (SIM) and the base unit handy (see "Locating the serial number" in the *HST-3000 Base Unit User's Guide*).

Table 1 Technical assistance centers

Region	Phone Number	
Americas	1-866-ACTERNA 1-866-228-3762 301-353-1550	tac@jdsu.com
Europe, Africa, and Mid-East	+49 (0) 7121 86 1345 (JDSU Germany)	hotline.europe@jdsu.com
Asia and the Pacific	+852 2892 0990 (Hong Kong) +8610 6833 7477 (Beijing-China)	

During off-hours, you can request assistance by doing one of the following: leave a voice message at the TAC for your region; email the North American TAC (tac@jdsu.com); submit your question using our online Technical Assistance request form at www.jdsu.com.

Conventions

This guide uses conventions and symbols, as described in the following tables.

Table 2 Typographical conventions

Description	Example
User interface actions and buttons or switches you have to press appear in this **typeface**.	Press the **OK** key.
Code and output messages appear in this `typeface`.	`All results okay`
Text you must type exactly as shown appears in this **`typeface`**.	Type: **`a:\set.exe`** in the dialog box.
Variables appear in this **typeface**.	Type the new **hostname**.
Book references appear in this *typeface*.	Refer to *Newton's Telecom Dictionary*

Table 3 Keyboard and menu conventions

Description	Example
A plus sign + indicates simultaneous keystrokes.	Press **Ctrl+s**
A comma indicates consecutive key strokes.	Press **Alt+f,s**
A slanted bracket indicates choosing a submenu from menu.	On the menu bar, click **Start > Program Files**.

Table 4 Symbol conventions

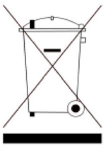
	This symbol represents a general hazard.
	This symbol represents a risk of electrical shock.
	This symbol represents a risk of explosion
	This symbol represents a Note indicating related information or tip.
	This symbol, located on the equipment, battery, or packaging indicates that the equipment or battery must not be disposed of in a land-fill site or as municipal waste, and should be disposed of according to your national regulations.

Table 5 Safety definitions

DANGER	Indicates an imminently hazardous situation which, if not avoided, will result in death or serious injury.
WARNING	Indicates a potentially hazardous situation which, if not avoided, could result in death or serious injury.
CAUTION	Indicates a potentially hazardous situation which, if not avoided, may result in minor or moderate injury.

Getting Started

1

This chapter provides a general description of the HST-3000's optional VoIP testing features. Topics discussed in this chapter include the following:

Overview and options

The HST-3000's optional VoIP testing features (ordering number HST3000-VOIP) allow placing and receiving calls, measuring call quality, and tracing the call route.

The capabilities of the HST-3000 include the following:

- Call setup and teardown
- Headset or speaker/mic
- Voice conversation/generate tone/IP voice announce
- Auto answer
- Real-time packet metrics (delay, jitter, packet loss)
- E-model QoS and RTCP statistics
- User selectable CODEC
- Ethernet testing
  - Physical layer statistics
  - Ping capability
  - FTP/HTTP throughput
  - Trace route

You can expand your testing capability by purchasing additional VoIP testing options. The options available for purchase are as follows:

Table 1 VoIP testing options

Option	Description	Order Number
H.323 call control	VoIP Signalling call controls for H.323	HST3000S-H.323
SCCP call control	VoIP Signalling option for Cisco SCCP	HST3000S-SCCP
SIP call control	VoIP Signalling option for SIP	HST3000S-SIP
MOS analysis	Mean Opinion Score (MOS) analysis option	HST3000S-MOS

Table 1 VoIP testing options (Continued)

Option	Description	Order Number
MGCP call control	Allows testing with Media Gateway Control Protocol call control	HST3000-MGCP
UNIStim call control	Allows testing with UNIStim call control	HST3000-UNISTIM
Video conferencing	Allows analysis of video packets for video conferencing applications that use SIP call control.	HST3000-VIDEO-CONF
Secure VoIP	Allows encrypted VoIP calls	HST3000-SECUREVOIP

What's new in this release

The following features were added in this release.

- Monitor mode.
  In Monitor mode, the HST monitors up to five VoIP calls between other phones and analyzes the quality of the RTP stream between the phones. For more information, see "Monitoring VoIP calls" on page 42.
- Support for Unistim secure RTP/RTCP (encrypted audio) and SIP re-invite.

Quick tour

The following section describes the status indicators and connections applicable to VoIP.

Status LEDs

These indicators report the status of the application. The function of each LED is described in Table 2.

Table 2 Status LEDs

LED	Function
Sync	A two-color LED used to indicate synchronization on the transport layer (for example, ADSL or Ethernet). **For xDSL:** – Flashing green indicates that the modems are training. – Solid green indicates that the modems have synchronized. – Solid red indicates a synchronization error has occurred. **For Ethernet:** – Solid green indicates 10/100 activity has been detected.
Data	A two-color LED that reports the status of the data connection. – Flashing green indicates that the data connection is not yet established. – Solid green indicates that a data connection has been established with the network (so that the HST-3000 may send and receive data on the network). In DHCP mode, it indicates that we have achieved an IP address assignment. In Static IP mode, it indicates that 10/100 activity has been detected. – Solid red indicates a data connection error. – If no data link is present, the Data LED will be dark.
Error	A red LED that indicates a gatekeeper error.
Alarm	A two-color LED that reports quality of service (QoS). – Amber indicates poor QoS. – Red indicates QoS failed.
LpBk	The loop back LED is not used for VoIP.
Batt	Indicated battery status. For more information, see the *HST-3000 Base Unit User's Guide*.

Connections

The HST-3000 top panel, is used to connect test leads, an audio headset, or to connect to an external output device. Figure 1 shows the top panel.

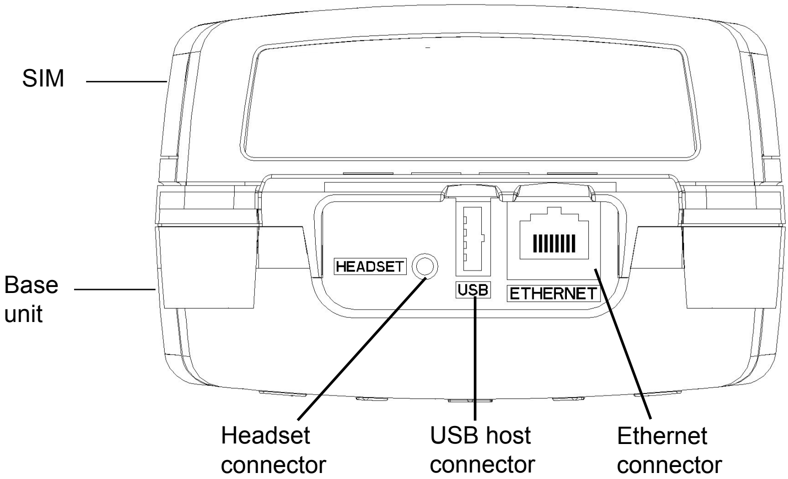


Figure 1 HST-3000 top panel

The connectors on the top panel are described in the following sections.

Headset connector

The Headset connector is used to connect a 2.54mm microphone/speaker audio headset.

USB connector

This USB host connector is used to connect the HST-3000 to a printer (device) for transferring test statistics.

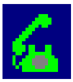
Ethernet connector

The Ethernet connector is used to connect a 10/100 BaseT Ethernet cable. If connecting to a PC, a cross-over cable should be used.

Status icons

Offhook and onhook icons appear in the upper right corner of the menu bar indicating the state of the phone. Table 3 describes these status icons. For descriptions of other status icons, see the *HST-3000 Base Unit User's Guide*.

Table 3 Status icons

Icon	Function
	Indicates the phone is onhook.
	Indicates the phone is ringing.
	Indicates the phone is offhook.

User interface navigation

You can use the Home and Config keys to cycle through the applications. The menus cycle in order, top to bottom as shown on the main application screen.

For example, if you are testing on an ADSL interface using an ADSL SIM and you are in the Modem Emulate application, pressing Home will move to the Data application, and then the VoIP application, and then IP Video.

The same methodology applies to the configuration menus. If you are viewing the General modem settings, pressing Config will move to the Data settings, and then VoIP General settings, and then the Video Settings.

VoIP Testing

2

This chapter provides task-based instructions for using the optional HST-3000 voice over IP (VoIP) testing features. Topics discussed in this chapter include the following:

Accessing the VoIP testing feature

Using the HST-3000 VoIP testing features, you can perform the following tasks:

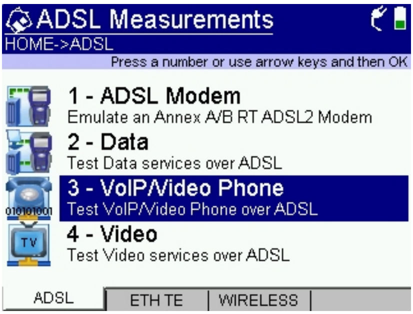
- Place and receive VoIP calls
- Emulate IP terminal equipment
- Trace the call route
- Capture and save VoIP packets to a file
- Monitor VoIP calls

The following procedure describes how to access the VoIP testing feature.

To access the VoIP testing feature

1 From a main measurements page, select **VoIP**.

For example, from the ADSL Measurements menu, press the **3** key.



If your unit is equipped with the video conferencing option, the item name will be “VoIP/Video Phone”. (The menu above indicates the option is installed.)

NOTE:

If you are testing VoIP over xDSL, the Data mode must be set (in the Data application, from the Configuration menu). If it is off, VoIP testing is disabled.

The VoIP Phone menu appears.

The following sections describe how to use the VoIP testing features.

Specifying test settings

Before you begin testing, make sure that the HST-3000 settings match the settings of the phone you are emulating. The following sections describe how to specify the settings.

> **NOTE:**
>
> You must specify the interface settings before specifying VoIP settings. For example, you must configure the Data, LAN, and other applicable settings for a xDSL line.

Specifying general settings

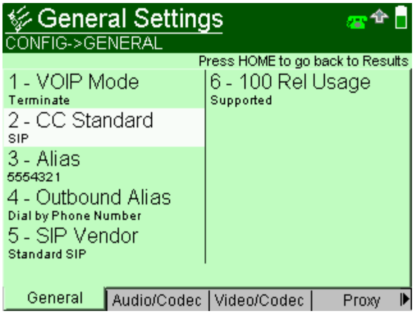
The following procedure describes how to specify the general settings.

To specify the general settings

1 Press the **Configure** navigation key.

2 Press the **General** soft key.

The General Settings menu appears.



3 Select **VOIP Mode** and specify whether you are terminating or monitoring.

– **Terminate** mode allows the HST to act as a VoIP phone.

– **Monitor** mode allows the HST to monitor VoIP calls between other phones and analyze the quality of the RTP stream between the phones.

4 If you selected Monitor mode, specify the following settings.

Setting	Description
Transport Type	Specifies what type of network traffic to monitor.
Use VLAN Filter?	Specifies whether to monitor only traffic from a specific VLAN.
VLAN Value	If Use VLAN Filter is “Yes”, this specifies which VLAN to monitor.
Use IP Filter?	Specifies whether to monitor only traffic from a specific IP address.
IP Value	If Use IP Filter is “Yes”, this specifies which IP address to monitor.

5 If you selected Terminate mode, select **CC Standard**, and then specify a call control standard.

> **NOTE:**
>
> The call controls are options. Your selections will vary depending on which call control options are installed on your unit.

- **H.323 (Fast connect)** minimizes the number of messages exchanged. Not all equipment supports this option.
- **H.323 (Full)** provides a full suite of H.255 and H.245 message exchange with the far end, including negotiation of codec and capabilities.
- **SIP** is Session Initiation Protocol. It is an application layer protocol used to establish, modify, and terminate conference and telephony sessions over IP-based networks.
- **SCCP** is the call control used on Cisco VoIP systems.
- **MGCP** is Media Gateway Control Protocol.
- **UNISTIM** is used on Nortel systems.

6 If you selected H.323 call control, specify the following settings.

Setting	Description
Alias	Enter an alias address. This is the phone number alias that will be associated with the phone that you are trying to emulate. This phone number will be included in the connection request messages that are exchanged with the Gatekeeper or other endpoint devices

Setting	Description
H.323ID	Enter the ID, using up to 40 characters. This is an ID element field in the ALIAS information that is sent to the Gatekeeper during all registration and request messages. The H.323 gateway registers with an H.323 ID. This is either an email ID or an E.164 address. For example: Email ID (H.323 ID): gwy-01@domain.com E.164 Address: 5125551212
Bearer Cap	Specify the bearer capability: Voice, 3.1K audio, Unrestricted digital This sets the Bearer Cap information element in the H.323 setup message for outgoing calls.
Calling Plan	Specify the numbering plan, if required: Unknown, ISDN/Telephony, Data, Telex, National, Private This sets the Calling Party Numbering Plan information element in the H.323 setup message for outgoing calls.
Calling Type	Specify the type of number, if required: Unknown, International, National, Network Specific, Subscriber, Abbreviated. This sets the Calling Party Type information element in the H.323 setup message for outgoing calls
Called Plan	Specify the numbering plan, if required: Unknown, ISDN/Telephony, Data, Telex, National, Private. This sets the Called Party Numbering Plan information element in the H.323 setup message for outgoing calls.

Setting	Description
Called Type	Specify the type of number, if required: Unknown, International, National, Network Specific, Subscriber, Abbreviated. This sets the Called Party Type information element in the H.323 setup message for outgoing calls.

7 If you selected SIP call control, specify the following settings.

Setting	Description
Alias	Enter an alias address. This specifies the registrar. If you have a network that uses one server for registration (the registrar) and another for placing and receiving calls “Alias” specifies the registrar. (“Proxy IP” on the Proxy Settings menu specifies the server used for placing and receiving calls.) This phone number will be included in the connection request messages that are exchanged with the Proxy or other endpoint devices.
Outbound Alias	Select one of the following options: Dial by Phone Number or Dial by Name/URL/Email.
SIP Vendor	Specify the vendor.
100 Rel Usage	Specify whether 100rel is required, supported, or disabled. 100 Rel provides reliable provisional response messages by appending the 100rel tag to the value of the required header of initial signalling messages.

8 If you selected SCCP call control, specify the type of device and the device name:

To...	Do the following
Specify the type of device	Select **Device Type**, and then specify the device type.
Specify the type of name for the device	Select **Device Name Type**, and then select one of the following: – Automatic based on MAC address – User Defined
Specify the name of a device	Select **Device Name**, and then enter a name. **NOTE:** The Device Name option is only available if Device Name Type is configured to “User Defined.”

9 If you selected UNISTIM call control, specify the following settings:

Setting	Description
DHCP Provisioning	Specify whether DHCP Provisioning is enabled. If enabled, an extra option string is sent during the call setup that tells the far-end phone where to find the server, what port to use, and so on. (It also changes the following on the Data Settings: *Use Vendor ID* set to “Yes” and *Vendor ID* set to “Nortel-i2004A”.)
Phone Type	Specify the type of phone.

Setting	Description
Prod Eng Code	Specify the code. The Product Engineering Code is a number in the setup message that identifies the type of phone. If you find that there is a mismatch in what the switch is expecting and the default code for that phone type, enter the expected code. For example, if you specified an i2004 phone but the code identified it as an i2006, which may happen when you have a newer phone on an older system or vice-versa, you can enter a different code for that phone.
Verbose Output	Specify whether detailed screen-based text messages will be displayed (On). The MCS prompts only selection will display only Multimedia Communication Server prompts, or prompts/text that the switch told the unit to write to the screen, such as “enter password.”

10 If you selected MGCP call control, specify the following settings.

Setting	Description
Endpoint ID Type	Specify the **Endpoint ID Type**. The type can be user defined or automatic based on the IP address (for example, aaln/01@[10.50.20.2])
Endpoint ID	If you selected User Defined as the type, enter the Endpoint ID.

The general settings are specified.

Specifying Audio/Codec settings

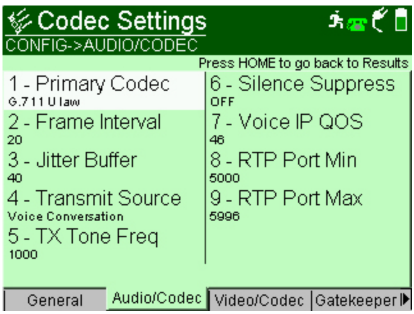
The following procedure describes how to specify the Audio/Codec settings.

To specify the Audio/Codec settings

1 Press the **Configure** navigation key.

2 Press the **Audio/Codec** soft key.

The Audio/Codec Settings menu appears.



3 Specify the following settings:

Setting	**Description**
Primary Codec	This selects which codec will be used. (Not available if using UNIS-TIM call control)
Frame Interval	Set the frame interval. This is the number of milliseconds of speech per transmission frame when using a sample based codec (such as G.711).
jitter buffer	Set the jitter buffer length.This is the number of milliseconds of speech that will be collected before an attempt will be made to play the speech back. This allows lost, late, or out-of-sequence packets time to arrive and be reassembled before playback.
Transmit Source	Select the transmit source: Voice conversation (transmits and receives live voice), IP voice announce (the unit repeats a sequence of words including the calling party's IP address), Tone (transmits the specified frequency).
Transmit Frequency	Enter the transmit frequency
Silence Suppression	Specify whether silence suppres-sion is on or off.

Setting	Description
Voice IP QOS	Enter a value to indicate the Voice IP Quality of Service. The value you enter will be both the Differentiated Services (DiffServ) code point and the type of service (ToS) indicator. The value will occupy a 6-bit field in the packet headers of RTP stream voice packets and will indicate how packets are treated at each hop. You can specify a number from 0 to 63 to indicate the per-hop behavior.
RTP Port Min/Max (all call controls except Unistim)	Specify the RTP port minimum and maximum numbers. The real-time transport protocol (RTP) port number allows you to identify voice traffic versus other traffic. Some systems only accept RTP traffic on certain port numbers.
DTMF In-Band (Unistim call control only)	Specify whether DTMF tones are sent in-band or out-of-band.

The Audio/Codec settings are specified.

Specifying Video/Codec settings

The following procedure describes how to specify the Video/Codec settings. This applies to the SIP call control only.

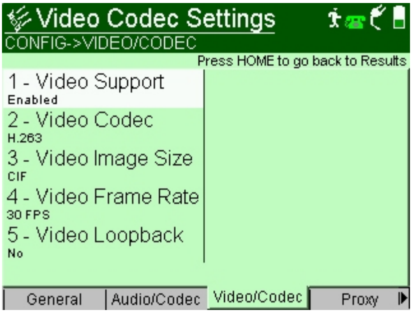
To specify the Video/Codec settings

1 Press the **Configure** navigation key.

2 Verify that you are using SIP call control (on the General settings).

3 Press the **Video/Codec** soft key.

The Video/Codec Settings menu appears.



4 Specify the following settings:

Setting	Description
Video Support	Select whether video support is enabled for the call. **NOTE:** If the value under Video Support is “Not Available”, you are using a call control that does not support video. See step 2. If you selected “No,” you are finished.
Video Codec	Specify the Video Codec: H.261 complies to ITU document H.261. H.263 (the default) complies to ITU document H.263.
Video Image Size	Set the resolution of the video: CIF is 288 x 352, QCIF is 144 x 176
Video Frame Rate	Specify the maximum frame transmission rate (frames per second): 30 FPS is full rate, 15 FPS is $^1/_2$ rate 10 FPS is $^1/_3$ rate 7.5 FPS is $^1/_4$ rate

Setting	Description
Video Loopback	Specify whether to loopback the video being received. This setting applies whether the HST-3000 initiates the call or receives the call. If Video Loopback is ON, the HST-3000 does not transmit any patterns (the color bars); it waits to receive video and then loops it back.

The Video/Codec settings are specified.

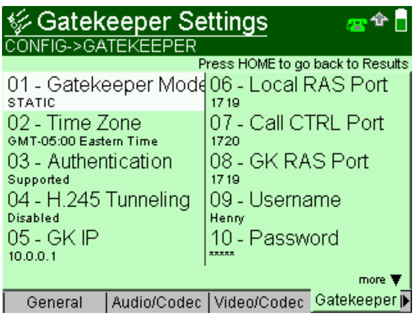
Specifying gatekeeper settings

The following procedure describes how to specify the gatekeeper settings. These settings apply to the H.323 call controls. If you selected SIP, SCCP, or MGCP call control, you will specify proxy or call manager/agent settings instead of the gatekeeper settings. See "Specifying proxy settings" on page 23 or "Specifying call manager/agent settings" on page 24. If you selected UNISTIM call control, you will specify TPS settings. See "Specifying TPS settings" on page 22.

To specify the gatekeeper settings

1. Press the **Configure** navigation key.
2. Press the **Gatekeeper** soft key.

   The Gatekeeper Settings menu appears.



3 Specify the following settings:

Setting	Description
Gatekeeper Mode	Specify the gatekeeper mode: **NO GATEKEEPER** means no RAS (registration, admission, and status) messages will be used. **AUTO DISCOVER** uses a well known IP address and port number. **STATIC** allows you to enter the gatekeeper address.
Time Zone	Select the time zone where you are located.
Authentication	Specify whether authentication is required, supported, or disabled.
H.245 Tunneling	Specify whether H.245 tunneling is enabled or disabled. H.323 version 2 and above support the tunneling of H.245 messages. To avoid the need to open a second TCP/IP channel for H.245, the H.245 message is encoded and inserted to any Q.931/H.225.0 message that is currently being sent.
GK IP	Enter the gatekeeper IP address
Local RAS Port	Enter the UDP port that is used locally for registration (RAS messages)
Call CTRL Port	Enter the UDP port that is used for call control messages (for placing and receiving calls).
GK RAS Port	Enter the UDP port that the gatekeeper uses for registration (RAS messages).
Username	Enter the username to register with the gateway.

Setting	Description
Password	Enter the password to register with the gateway.
GK ID	Enter the Gatekeeper ID, up to 20 characters.

The gatekeeper settings are specified.

Specifying TPS settings

For UNISTIM call control, the Terminal Proxy Server (TPS) settings must be specified.

To specify TPS settings

1. Press the **Configure** navigation key.
2. Press the **TPS** soft key.
   The TPS Settings menu appears.
3. Specify the following settings:

Setting	Description
TPS IP	Enter the TPS IP address
TPS Port	Enter the TPS Port number
IT Port	Enter the internet terminal (IT) Port.
S1 Action	Specify the S1 Action: Select **1 Standard Unistim** to set the bit to 1 and use typical Unistim protocol. Select **6 Secure Unistim (Fresh Key)** to set the bit to 6 and use secure Unistim using a new/fresh key code. Select **7 Standard Unistim (Restart Allowed)** to set the bit to 7 and use secure Unistim with a previous key code.

The TPS settings are specified.

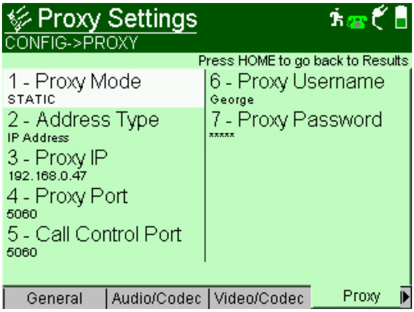
Specifying proxy settings

If you configured the call control standard to SIP (see "Specifying general settings" on page 9), you must also specify the proxy settings.

To specify the proxy settings

1 Press the **Proxy** soft key.

The Proxy Settings menu appears.



2 Specify the following settings:

Setting	Description
Proxy Mode	Select the Proxy Mode: STATIC or DHCP
Address Type	Select the Address Type: IP Address or DNS Name
Proxy IP	Enter the IP address of the proxy. This is the outbound proxy, or the device from which the HST will send and receive all SIP messages. If you have a network that uses one server for registration and another for placing and receiving calls, the Proxy IP specifies the address for placing and receiving calls ("Alias" on the General Settings menu specifies the registrar address).

Setting	Description
Proxy Port	Enter the proxy port number.
Call Control Port	Enter the call control port number.
Proxy User name	Enter a user name used to access the Proxy.
Proxy Password	Enter the Proxy password.

The proxy settings are specified.

Specifying call manager/agent settings

If you configured the call control standard to SCCP or MGCP (see “Specifying general settings” on page 9), you must also specify the call manager or call agent settings.

If using SCCP call control, the menu title is Call Manager; for MGCP, it’s Call Agent.

To specify the call manager/agent settings

1 Press the **Call Mnger** or **Call Agent** soft key.

The Call Manager (or Agent) Settings menu appears.

2 Select **C. Manager IP** or **Call Agent IP**, and then enter the IP address of the call manager or call agent.

3 Select **C. Manager Port** or **Call Agent Port**, and then enter a number for the call manager port. The range is from 1 to 65535.

The call manager/agent settings are specified.

Specifying QoS settings

The quality of service (QoS) measurement uses two thresholds (good and poor) to produce three quality ratings: good, poor, and fair. If the value is less than the good threshold, it is considered good and is indicated with a check mark. A value between good and poor is considered fair and is indicated by an exclamation mark. If the value is greater than the poor threshold, it is considered poor and is indicated by an X.

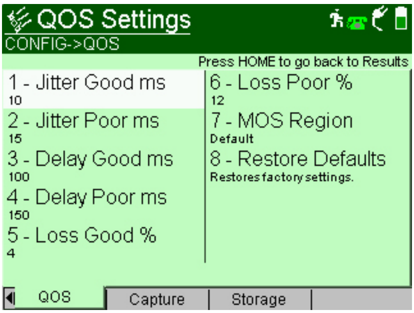
This menu defines the results thresholds for the QoS measurements.

To specify QoS settings

1 Press the **Configure** navigation key.

2 Press the **QOS** soft key. You nay need to press the right arrow key one or more times to find the QOS soft key.

The QOS Settings menu appears.



3 Specify the following settings:

Setting	Description
Jitter Good	Enter the Pass threshold, in milliseconds.

Setting	Description
Jitter Poor	Enter the Fail threshold, in milliseconds.
Delay Good	Enter the Pass threshold, in milliseconds.
Delay Poor	Enter the Fail threshold, in milliseconds.
Loss Good	Enter the Pass threshold, in milliseconds.
Loss Poor	Enter the Fail threshold, in milliseconds.
MOS Region	Select a **MOS Region**. This selection affects the scaling of the CQ-MOS and LQ-MOS results (on the "Call Scores" result screen). This selection is only available if your unit is equipped with the MOS option.

4 If you want to want to reset all the QoS VOIP defaults, select Restore Defaults.

The QoS settings are specified.

VoIP Testing

The HST-3000 supports multiple signaling protocols to emulate a VoIP phone. The VoIP phone test involves the following steps:

– Selecting phone mode
– Specifying test settings
– Connecting to the line
– Placing and receiving calls

Selecting phone mode

The first step for the test is to select phone mode.

To select phone mode

1. Select VoIP from the main menu (see page 8).

   The VoIP Phone or Video Phone menu appears.

   Video Phone appears if your unit is equipped with the video conferencing option.

2. If the Phone menu is not displayed, press the **Display** soft key then the **1** key to select Phone mode.

   The VoIP Phone menu appears.

Specifying test settings

After selecting the phone mode, the test settings must be specified. See "Specifying test settings" on page 9.

Connecting to the line

After specifying the test settings, you can connect to the line.

To connect to the line

1. For Ethernet connections, do the following:

   a. Connect one end of an Ethernet cable to the Ethernet jack on the top of the unit.

   b. Connect the other end of the cable to an Ethernet jack. If you are connecting to a PC, use a cross-over cable.

2. For DSL connections, do the following:

   a. Connect one end of the DSL cable to the DSL jack on the right side of the SIM.

   b. Connect the other end of the cable to either the NID (demarc) or the wall jack.

You are connected to the line.

If you are placing or receiving calls, we recommend that you also connect a headset to the headset jack.

Placing a call

After specifying test settings and connecting to the line, press the **Home** navigation key to return to the VoIP Phone menu. The information that appears on this menu varies depending on the gatekeeper mode that you specified in the Configuration menu.

If your gatekeeper mode is Auto or Static, the VoIP phone menu appears as shown in Figure 2.

Figure 2 VOIP Phone menu with a gatekeeper/proxy

If your gatekeeper mode is None, the VoIP phone menu appears as shown in Figure 3.

Figure 3 VOIP Phone using H.323 and no gatekeeper

The following procedure describes how to place a VoIP call.

To place a VoIP call

1 Select one of the following options:

If...	Then...
The gatekeeper mode is Auto or Static	– To enter a number using speed dial, press the **Speed** soft key, and then select an entry from the phone list (For more information about the phone book, see “Using speed dial” on page 30). – To enter a number manually, press the 1 key and then enter the a number. Use the **Delete** and **Clear** soft keys to erase numbers.
The gatekeeper mode is None	– Press the **1** key. – Enter the phone number – Press the **OK** key. – Press the **2** key. – Enter the IP address. – Press the **OK** key.

2 Press the **OK** key to go off hook and place the call.

After the call is connected, the quick reference results will update in the bottom right area of the screen.

3 To view additional results, press the **Display** soft key, and then select a result category.

> **NOTE:**
>
> You can also use the left or right arrow keys to move to different result categories.

4 Press the **Results** soft key to clear or save the results (see “Saving results” on page 40).

5 Press the **OK** key to go back on hook and end the call.

See Chapter 3 “Interpreting Test Results” on page 45 to learn what your results mean.

Using speed dial

The speed dial feature gives you quick access to phone numbers when you want to place a call. You can add, edit, and delete speed dial entries on the HST-3000. Entries can include name, phone number, and IP address information. The entries are saved when you turn off the HST-3000.

The following procedures describe how to add, edit, and delete phone book entries.

Adding a speed dial entry

Adding a speed dial entry allows you to save a phone number and/or IP address to the HST. This allows you to quickly enter the information when you use the HST to place a call.

To add a speed dial entry

1. From the VoIP Phone menu, press the **Speed** soft key.

   The Manage Entries option appears. If you saved speed dial entries previously, the saved entries also appear.
2. Select **Manage Entries**, and then select **Add**.
3. Specify a name for the entry.
4. Enter a phone number alias for the entry.
5. Enter an IP address, if needed.
6. Press the OK key.

   The entry is saved.
7. To view the new entry, press the **Speed** soft key.

   The entry appears in the list of saved numbers.

The speed dial entry is added.

Deleting a speed dial entry

Deleting an entry removes it from the speed dial list.

To delete a speed dial entry

1. From the VoIP Phone menu, press the **Speed** soft key.

The Manage Entries option appears. If you added speed dial entries previously, the existing entries also appear.

2. Select **Manage Entries**, and then select **Delete**.

   A list of entries appears.

3. Select the entry you want to delete.

The entry is deleted from the speed dial list.

Editing a speed dial entry

To Edit a speed dial entry

1. From the VoIP Phone menu, press the **Speed** soft key.

   The Manage Entries option appears. If you added speed dial entries previously, the existing entries also appear.

2. Select **Manage Entries**, and then select **Edit**.

   A list of entries appears.

3. Select the entry you want to edit.
4. Change the entry name, and then press the **OK** key.
5. Change the phone number alias, and the press the **OK** key.
6. Change the IP address, and then press the **OK** key.

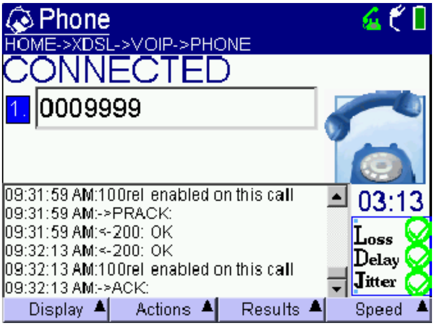
The edited entry is saved to the HST.

Receiving a call

The following procedure describes how to receive a VoIP call.

1. When the HST-3000 signals an incoming call, press the **OK** key to go off hook and answer the call.

After the call is connected, the quick reference results will update in the bottom right area of the screen.



A check mark indicates pass; an "X" indicates fail; an exclamation point indicates a marginal result.

2 To view additional results, press the **Display** soft key, and then select a result category.

> **NOTE:**
>
> You can also use the left or right arrow keys to move to different result categories.

3 Press the **Results** soft key to clear or save the results (see "Saving results" on page 40).

4 Press the **OK** key to go back on hook and end the call.

See Chapter 3 "Interpreting Test Results" on page 45 to learn what your results mean.

Answering calls automatically

The Auto Answer feature allows you to verify incoming service as well as caller ID service. The following procedure describes how to answer calls automatically.

To answer calls automatically

1 Access the VoIP testing feature (see page 8).

2 Press the **Actions** soft key.

3 Select **Turn Auto Answer On**.

4. Place a call to the HST from a VoIP phone (or a second HST).

   The HST answers the call with a voice announcement, and then the quick reference results will update in the bottom right area of the screen. If the codec does not support IP voice announce, a tone is transmitted.

   A check mark indicates pass; an “X” indicates fail; an exclamation point indicates a marginal result.

5. Press the **Clear** soft key to clear the log.

6. To exit Auto Answer mode, press the **Actions** soft key, and then select **Turn Auto Answer Off**.

Analyzing video conference calls

If you are using the SIP call control, the HST-3000 supports video conferencing — the capability to pass video as well as audio during a VoIP call. In this mode, the HST-3000 does not actually display video, it analyzes the packets that contain video, as well as analyzing the audio packets.

Call negotiation

When a call is initiated, the HST-3000 negotiates support with the far end for the codec, size, and frame rate you specified, and, assuming successful negotiation, will connect the call.

For the call to be successful, both ends of the call must support video conferencing and use the same codec. If either of these don’t match, the call will not connect. For the image size and frame rate, the negotiation will result in the smaller of the two.

Loopback After the call is connected, transmission from the HST depends on the Video Loopback setting you specified (on the Video/Codec Settings screen).

If Video Loopback is Off, the HST-3000 transmits specific video test patterns (colored bars) with different codec/resolution/frame rate combinations. Other sources of video, such as cameras, are not supported.

If Video Loopback is On, the HST-3000 simply loops back the video it receives.

The following procedure describes how to analyze video conference calls.

To analyze video conference calls

1. If you haven't already done so, do the following:
   a. Specify the Video/Codec settings (see "Specifying Video/Codec settings" on page 18).
   b. Connect to the line.
2. Place a call.
3. View the results in the following categories.
   - QoS
     – Audio
     – Video
   - Call Scores
     – Simple Overall MOS/VQS
     – Current Call Details
     – Historical Call Details
     – Audio Degradation Factors

   See Chapter 3 "Interpreting Test Results" on page 45 to learn what your results mean.

You have analyzed a video conference call.

Emulating a UNISTIM phone display

If you are using the UNISTIM call control, and registered with the server, the HST-3000 can emulate the display for the type of phone you specified on the General Settings configuration menu. This is useful when registering a phone and logging onto a network. Figure 4 illustrates a typical phone display.

Figure 4 UNISTIM phone display example

To emulate a phone display

1 If you haven't already done so, do the following:

  a On the General Settings menu, specify the following:

  – Select UNISTIM as the CC Standard.

  – Select the Phone Type.

  b Connect to the line.

2 Press the **Home** navigation key to view the VoIP Phone menu.

3 Press the **Display** soft key and then press **2** to select the Unistim Phone display, for example "Nortel i2004".

4 Navigate through the display as needed:

  – Use the soft keys to select the features shown by the soft key labels on the phone.

NOTE:

The HST-3000's common soft keys (such as **Display**, **Actions**, and **Results**) are disabled on this screen.

- – Use the keypad for data entry (both numeric and text) when prompted by the server. However, the interpretation of the keystrokes is server dependent.
- – When a scrollable menu is displayed by the server, use the up and down arrow keys to scroll a menu up and down.
- – When character backspacing is allowed by the server, use the shift+left arrow sequence to backspace one character.
- – Use the OK key to toggle on hook and off hook.

5 To exit, use the left or right arrow key to display another category of results.

Capturing VoIP packets

The HST-3000 allows you to capture messages (packets) exchanged during a VoIP call to a file on the HST or to a file on a USB drive.

The capture feature allows you to filter captured information by IP address, port number, or by a user-defined filter string. The information can be saved to a file, using PCAP format, which can be sent to a network technician for analysis on a protocol analyzer.

If you turn on the HST-3000 with a USB drive attached, the HST-3000 will recognize it and the USB drive will become the default location where the packets will be saved. Otherwise, the packets will be saved to a file in the internal File Manager. The maximum file size for packets stored in the internal File Manager is 1 Mbyte; for USB capture, file size depends on the space available on the USB drive.

The following sections describe how to specify the packet capture settings, enable/disable the packet capture feature, and how to save captured information.

Specifying packet capture settings

Before you capture packet information, you must specify the capture settings. You can specify the size of the capture file and set filters to refine the information.

To specify packet capture settings

1. Press the **Configure** navigation key.
2. Press the **Capture** soft key.

   You may have to use the left or right arrow key to locate the Capture soft key.
3. Press the **1** key to enable or disable Promiscuous mode.

   Enabling Promiscuous mode causes the HST to capture all IP traffic regardless of source or destination addresses.
4. Press the **2** key, and then specify the **Packet Limit**.

   The packet limit indicates how much information can be stored in the capture file (in other words, the size of the capture file). You can specify from 1 to 10000000 packets. When the maximum file size has been reached, the HST automatically stops capturing packets.
5. To filter messages by IP address, do the following:

   a. Press the **3** key, and then select **Yes** to enable the filter or **No** to disable the filter.

   b. If you enabled the filter, select **Filter IP**, and then enter an IP address.

      Only information from the IP address you entered will be captured.
6. To filter messages by port, do the following:

   a. Select **Use Port Filter**, and then select **Yes** to enable the filter or **No** to disable the filter.

   b. If you enabled the filer, select **Port Number**, and then enter a port number.

      You can enter a number from 1 to 65534. Only messages from the port you entered will be captured.

7 To define your own filter, do the following:

   a Select **Use Custom Filter**, and then select **Yes** to enable the filter or **No** to disable the filter.

   b If you enabled the filter, select **Custom String**, and then enter a text string using the HST keypad. Some commonly used strings are as follows:

String	Description
tcp	Captures only TCP packets
udp	Captures only UDP packets
igmp	Captures only IGMP packets
not tcp	Captures all packets except TCP
less 1000	Captures all packets less than 1000 bytes in length.
greater 1000	Captures all packets greater than 1000 bytes in length.

8 When you have finished specifying the capture settings, press the **Home** navigation key to return to the VoIP Phone screen.

You have finished specifying the packet capture settings. For information about starting packet capture, see "Enabling and disabling packet capture" on page 38.

Enabling and disabling packet capture

Before you begin capturing data, make sure the capture settings are set appropriately. See "Specifying packet capture settings" on page 37.

To enable or disable the packet capture feature

1 Place or receive a VoIP call. See "Placing a call" on page 28 or "Receiving a call" on page 31.

2 Select the **Actions** soft key, and then choose one of the following options:

– If packet capture was previously turned off, select **Start Packet Capture**.

A filename box appears. Enter a file name, and then press OK.

The HST begins to capture packets according to the specified capture settings.

If you have a USB storage device connected, it will be saved there. If not, the file will be saved to the HST in the /cust directory. For information on transferring files from the HST using FTP, see the *HST-3000 Base Unit User's Guide*.

CAUTION: CORRUPTED DATA

Removing the USB drive during a file capture or file save may corrupt the data. Do not remove the drive until all files have been saved or the packet capture is complete.

– If the capture feature was previously on, select **Stop Packet Capture**.

The HST stops capturing packets.

You have enabled or disabled the packet capture feature.

Saving results

In VoIP Phone mode, the HST allows you to save test results in five ways:

- To a new file
  - as an ASCII text file
  - in a tabular format, for graph results (tab separated values - can be used to import into a spreadsheet)
  - as a screen shot, for graph results
- Appended to an existing result file
- To a file at specific time intervals, for VoIP statistics

The following sections describe how to save results.

Saving to a file

The following procedures describe how to save results to a file.

To save results to a file

1 To save to an ASII text file, perform the following.

a From the VoIP Phone menu, press the **Results** soft key.

b Select **Save Results**.

A filename dialog box appears.

2 To save to a tabular file or as a screen capture, perform the following.

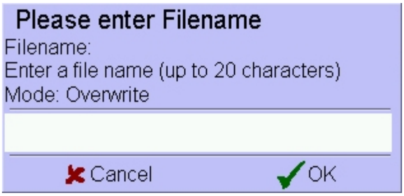
a From any QoS graph result screen, press the **Results** soft key.

b Select one of the following.

To save as...	Select...
Tab separated values	**Save Tabular**
Screen Capture	**Save Screen Capture**

A filename dialog box appears.

3 Enter a file name (up to 20 characters).



4 Press the **OK** key to save results to the file or press **Cancel** to exit.

The test results are saved to the file. For information about managing files, see the *HST-3000 Base Unit User's Guide*.

Appending result files

The following procedure describes how to save result data to an existing file without overwriting the file.

To append a result file

1 From the VoIP Phone menu, press the **Results** soft key.

2 Select **Save Results Append**.

3 Enter a file name (up to 20 characters).

You can enter the name of an existing file in which you want to save results, or you can enter a new file name.

4 Press the **OK** key to save results to the file.

The results are saved to the file. For information about managing files, see the *HST-3000 Base Unit User's Guide*.

Saving results at specific intervals

The following procedure describes how to save test results to a file at specific time intervals. For example, you can set the HST to save results to a file every 10 minutes.

To save results at specified intervals.

1 From the VoIP Phone menu, press the **Results** soft key.

2 Select **Start Timed Result Saving**.

3 Enter a file name (up to 20 characters), and then press the **OK** key.

   You can save results to an existing file, or you can enter a new file name.

4 Enter the time interval (in minutes), and then press the **OK** key.

   The interval save function is enabled. Test results will be saved to the file at the specified time interval. For information about managing files, see the *HST-3000 Base Unit User's Guide*.

5 To stop the interval save function, press the **Results** soft key, and then select **Stop Timed Result Saving**.

Monitoring VoIP calls

The VoIP Monitor allows the HST to monitor up to five VoIP calls between other phones and analyze the quality of the RTP stream between the phones.

> **NOTE:**
>
> The monitor feature only analyzes RTP packets; it does not analyze call control packets.

To monitor VoIP calls

1 Press the **Configure** navigation key.

2 From the General Settings menu, set the **VoIP Mode** to **Monitor**, and specify the test settings. For more information, see step 4 on page 10.

3 Connect to the line. See "Connecting to the line" on page 27.

4 Press the **Home** navigation key.

The VoIP Monitor screen appears.

If the HST locates a call stream, the IP address appears.

If Auto Analyze is On, the first received call will be answered automatically. If Auto Analyze is Off, calls must be answered manually.

5 Choose the call to monitor by highlighting the call, and then press the OK key.

The phone icon for the call will change to off hook and, if a video stream is detected, the icon changes to a TV.

For more information on the results, see "Monitor results" on page 47.

Using VoIP Inspector

The VoIP Inspector scans the circuit and reads the phone traffic to provide configuration information about the phone. This allows easier configuration of the HST for a particular phone.

> **NOTE:**
>
> VoIP Inspector is used with SIP and UniStim call controls only. In addition, it is not available during a call or in Monitor mode.

To use VoIP Inspector

1 Press the **Display** soft key and then select **Inspector**.

2 Do one of the following:

a Select Scan File.

– Select a capture file.

b Select Scan Live.

This selection appears if you are on an Ethernet circuit and you have an active data connection.

– Wait one minute.

– Select Stop Scan.

The VoIP Inspector results screen appears, showing the number of phones found (up to 5).

3 To clone a phone do the following:

a Use the keypad to select which phone to clone.

The configuration settings for that phone appear.

b Press the **OK** soft key to confirm that you want to use the configuration.

– If the phone requires a password, an “Enter Password” box appears.

– Enter the password, and then press the **OK** soft key.

The settings are loaded.

Interpreting Test Results

3

This chapter describes the test results that are gathered when running a test. The available test results vary depending on the mode. Topics discussed in this chapter include the following:

About the results

The following statistics are available:

– Phone
– “UNISTIM Phone” (for example, **Nortel i2004** or **Meridian M6350**)
– Monitor
– QoS
– Call Scores
– Throughput
– Call info
– Miscellaneous
– Delay

To navigate between the result screens, use the left and right arrow keys.

Phone results

This category lists events (and time stamps) that occur when you place or receive a call.

For quick reference, the Loss, Delay, and Jitter results appear in the lower right corner of this screen. If you are using SIP call control, the worst results for either the audio or video path will be displayed. For more information about these results see “QoS results”.

“UNISTIM Phone” results

If you are using UNISTIM call control, the second item on the list of results is the “UNISTIM Phone” results. The actual name of the screen varies, depending on the type of phone you specified on the General Settings configuration page. For example, if you selected Nortel i2004, the title for the result screen is “Nortel i2004.”

This screen emulates the phone display of the selected phone. This means that any text that would be displayed on the phone screen will be displayed on the HST-3000 screen.

If you are not registered with the terminal proxy server (TPS), the registration status appears.

Monitor results

When in Monitor mode, selecting a stream from this screen displays the general statistics for that stream, including the IP addresses and ports and codecs for each stream. Figure 5 shows a typical Monitor result screen.

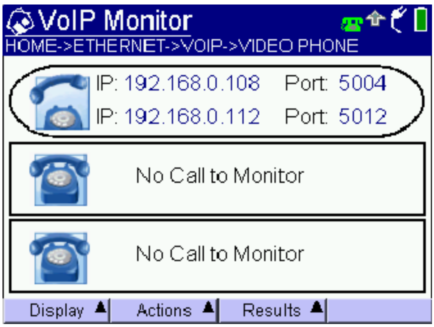


Figure 5 VoIP Monitor result example

QoS results

This category provides local and remote quality of service test results in both text and graphical formats, and are available for the Audio packets as well as the Video packets. Table 4 describes the QoS results.

Table 4 QoS results

Result	Definition
Delay	End to end delay in milliseconds
Jitter	Deviation in packet arrival times, in milliseconds
Loss	Packet loss based as a percentage lost divided by total packets (text only, no graph)
Overall	A rating (good, fair, poor) of the combination of delay, jitter, and loss

NOTE:

The Delay result is only calculated if the RTCP signaling is active.

Bandwidth is also available in graph form. It indicates the bandwidth used by that portion of the call: video section of bandwidth or audio section of bandwidth.

To switch among the text summary view and the graphs, press the **Results** soft key or the up or down arrow.

Figure 6 and 7 show typical graphic results for Delay and Jitter.

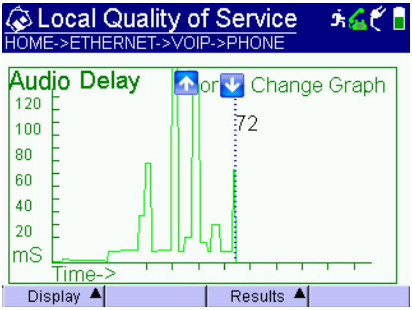


Figure 6 Typical Delay Graph

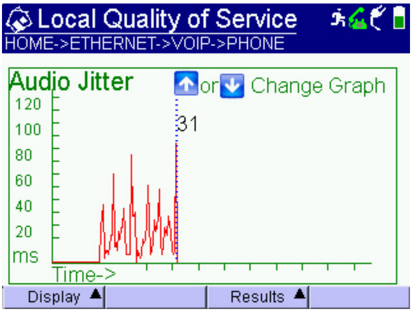


Figure 7 Typical Jitter Graph

For more information on the graphic results, see “Graphic displays” on page 57.

Call score results

When you select the Call Scores result category, you can view mean opinion scores and R factor results in three different ways: Simple Overall, Detailed Text, Degradation Factors.

To view call score results

1 Select the **Display** soft key.

2 Select **Call Scores**.

3 Press the **Result** soft key, and then select how to view the results:
   – **Simple Overall MOS/VQS**
   – **Current Call Details**
   – **Historical Call Details**
   – **Audio Deg. Factors**

   Descriptions of the results for each view are listed in the following sections.

   If in Monitor mode, the screens show statistics for both ends of the call.

Simplified overall view

Table 5 describes the overall call score measurements.

Table 5 Call score measurements

Result	Definition
MOS	Mean Opinion Score represented as a number and a graphic representation of quality.
R Factor	Listener and conversation quality R factor scores.
VSTQ	Video Service Transmission Quality. The ability of a network to transport IP video signals.

Current and historical call details

Table 7 describes the mean opinion score results provided on the call details screens. Current results are reported on the **Current Call Details**; average, minimum, and maximum results are reported on the **Historical Call Details** (current call details are not included).

Table 6 Mean opinion score results

Result	Definition
CQ R	Current, Average, Minimum, and Maximum conversation quality R factor

Table 6 Mean opinion score results (Continued)

Result	Definition
LQ R	Current, Average, Minimum, and Maximum listener quality R factor
G.107 R	Current, Average, Minimum, and Maximum G.107 R factor
Burst R	Current, Average, Minimum, and Maximum burst R factor
Gap R	Current, Average, Minimum, and Maximum gap R factor
CQ MOS	Current, Average, Minimum, and Maximum conversation quality mean opinion score
LQ MOS	Current, Average, Minimum, and Maximum listener quality mean opinion score
VSTQ	Current, Average, Minimum, and Maximum video service transmission quality score.

Audio degradation factors

This screen lists the maximum possible R factor, the maximum R factor encountered during the test, and the current R factor. It, in effect, shows the factors that contributed to a positive or negative shift away from the maximum R factor.

Throughput results

Table 7 describes the throughput results.If in Monitor mode, the screens show statistics for both ends of the call.

Table 7 Throughput results

Result	Definition
Rate	Bit rate transmitted (TX) or received (RX) for the audio and video portions of the call.

Table 7 Throughput results (Continued)

Result	Definition
Bytes	Total number of bytes transmitted (TX) or received (RX)
Packets	Total number of packets transmitted (TX) or received (RX)
Out of sequence	Total number of packets that arrive out of sequence
Remote Bytes TX	The number of Bytes that the remote entity reports that it has transmitted (sent via RTCP)
Remote Pack-ets TX	The number of Packets that the remote entity reports that it has transmitted (sent via RTCP)

Call stats results

This category provides results for the current call. Table 8 describes the call info results.

Table 8 Call stats results

Result	Definition
Call Duration	Length of time for the current call.
Far End IP	The IP address of the incoming call
Far End Name	The ID of the incoming call
Far End Alias	The alias of the incoming call
Audio Codec Rx	The current Audio Codec the HST is using
Audio Codec Tx	The Audio Codec the remote entity reports that it is using

Table 8 Call stats results (Continued)

Result	Definition
Audio RTCP Used	Indicates whether RTCP was used for the Audio path
Audio Silence Suppression	Indicates whether silence suppression was used for the audio path
Video Codec Rx (if using SIP)	The current Video Codec the HST is using
Video Codec Tx (if using SIP)	The Video Codec the remote entity reports that it is using
Video RTCP Used (if using SIP)	Indicates whether RTCP was used for the Video path

Miscellaneous measurements

Table 9 describes the miscellaneous results for the audio path.

Table 9 Miscellaneous results

Result	Definition
Mic signal level	The current input signal strength, measured in dBm
Speaker signal level	The current output signal strength, measured in dBm
Replay	Number of jitter buffer replays
Idle	Number of times the jitter buffer has been idle
Dropped	Number of times the jitter buffer has dropped cells

Delay results

Table 10 describes the delay results for the audio path.

Table 10 Delay results

Result	Definition
Network	Time, in milliseconds, needed to travel the network
Encoding	Time, in milliseconds, needed to convert samples in selected codec form
Packetization	Number of milliseconds needed to fill the frame(s) comprising one RTP data packet
Buffering	Time, in milliseconds, that the data was held in a jitter buffer
Total	Total of all delays

Call log

This category provides a running log of significant events and errors. For example, when a call started or ended, or when the QoS condition changed.

Troubleshooting

4

This chapter describes how to identify and correct problems related to VoIP testing with the HST-3000. You should verify whether your problem is listed here before contacting technical assistance. Topics discussed in this chapter include the following:

Placing and receiving calls

The following section addresses questions about placing and receiving calls.

Issue

The call didn't go through.

Resolution

Check your connections to verify that they are hooked up properly.

Check the Ethernet link light on the HST-3000 Ethernet jack. It should be green.

Verify the LAN settings (IP address, netmask, DNS name).

Verify the call control. Most equipment uses Fast Connect.

If you *do not* have a gatekeeper, verify the outgoing alias and IP address.

If you *are* using a gatekeeper, verify you are registered with the gatekeeper.

Check with your system administrator to verify that the firewall allows VoIP traffic.

Issue

I am emulating a SIP phone but cannot register with the SIP server.

Resolution

In typical networks, the same server handles both registration and placing and receiving calls. However, in some networks, there is a Proxy server that handles SIP messaging for placing and receiving calls, and a registrar that handles registration, which may be in a different domain.

If this is the case, do the following.

– Verify that you specified the “Proxy” on the Proxy Settings menu as the outbound proxy, or the device from which the HST will send and receive all SIP messages (for placing and receiving calls).
– Verify that you specified the “Alias” on the General Settings menu as the SIP server or registrar (the device that keeps track of all the registered devices), using the following format “phoneNumber@domain” where domain is either an IP address of the registrar or a literal domain such as “jdsu.com”.

Graphic displays

The following section addresses questions about the graphic displays of Loss, Jitter, and Delay.

Issue

The delay measurement does not appear.

Resolution

The delay measurement is only displayed if RTCP is supported.

Issue

I have very little loss, but a high level of delay.

Resolution

Check your network. It may be experiencing high traffic.

Issue

I have a large amount of jitter, but no loss or delay.

Resolution

Check the setup of your router.

Ethernet TE mode

The following section addresses questions about Ethernet TE mode.

Issue

The ping menu says pings are being sent, but the network statistics are not incrementing.

Resolution

Verify that the network is up by checking the Data LED (it should be solid green).

Check the Ethernet link light on the HST-3000 Ethernet jack. It should be green.

Verify the IP address and netmask.

You may be behind a firewall. Use trace route to determine whether this is the case.

The ping function only *attempts* to send a ping every second. Depending on certain conditions, a physical ping packet may not be sent.

If IPoE or PPPoE data mode is being used, the device has to ARP the address first. If this fails eventually you will see a ARP HOST UNREACHABLE message. Check to see that the destination IP address and your configured IP parameters are correct.

If the Ethernet interface is being used, make sure that the cabling is correct. If the Ethernet cable is not hooked up, or is hooked up incorrectly, a packet will not be sent. Thus the Ethernet statistics will not increment.

Issue

The Sync LED is lit, but the Data LED is not lit.

Resolution

If you are on an IPoE line and STUN is enabled, the data layer will not come up until the STUN client on the HST has determined the type of NAT used between the HST and the STUN server. Wait a few minutes and try to resync.

Glossary

A

Alias — An alternate method of addressing the endpoint. Alias addresses include private telephone numbers, public E.164 numbers, numeric strings representing names, e-mail like addresses, etc. Alias addresses are unique within a zone.

D

Delay — How long, usually in milliseconds, it takes data to move from source to destination. Total delay can be broken down into encoding, packetization, network transmission, and buffering sub-categories.

DHCP — Dynamic Host Control Protocol.

DNS — Domain Name Server.

DTMF — Dual Tone Multi-Frequency. A voice-band tone-based method of signaling.

E

Ethernet — One of the most common local area network (LAN) technologies. It operates over a variety of physical media at various speeds. Most commonly used are 10/100 Base-T, 10 Mbps, and 100 Mbps over twisted pair cable.

G

G.711 — An ITU-T codec standard that specifies a pulse code modulation (PCM) technique for handling audio data.

G.726 — An ITU-T codec standard that specifies an adaptive differential pulse code modulation (ADPCM) technique for handling audio data.

GK — Gatekeeper. A network device that provides two-way, real-time connections between endpoints.

GW — Gateway. A network device that provides address translation and network access for endpoints, gateways, and MCUs. Optionally, it may also provide bandwidth control and network topology information.

H

H.245 — ITU-T recommendation related to control protocol for multimedia communication. This standard/recommendation specifies syntax and semantics of terminal information messages as well as procedures to use them for in-band negotiation communication.

H.323 — ITU-T recommendation related to packet-based multimedia communications systems. This is an umbrella standard/recommendation that defines standards for multimedia traffic over Local Area Networks (LANs) that do not guarantee quality of service.

I

ICMP — Internet Control Message Protocol. The protocol used to handle errors and control messages at the IP layer. ICMP is actually part of the IP protocol.

Internet Protocol (IP) — The network layer protocol for the Internet protocol suite.

IP Address — The 32-bit (IP version 4) address assigned to hosts that want to participate in a TCP/IP Internet.

IPoE — Internet protocol over Ethernet.

IT — Internet terminal. The equipment from which you receive configuration information.

J

Jitter — The variance in packet arrival times at an endpoint.

L

LAN — Local Area Network. A limited distance (typically under a few kilometers or a couple of miles) high-speed network (typically 4 to 100 Mbps) that supports many computers.

LED — Light Emitting Diode. The lights indicating status or activity on electronic equipment.

Loss — The dip in signal level between two points on a network.

M

MGCP — Media Gateway Control Protocol.

N

Netmask — Network mask. Used to determine if an IP address is on your subnetwork.

P

PPPoE — Point to point protocol over Ethernet.

R

RAS — Registration, admission, and status.

Route — The path that network traffic takes from its source to its destination. The route a datagram may follow can include many gateways and many physical networks.

RTCP — Real Time Control Protocol.

S

SCCP — Signaling Connection Control Part. SCCP is part of the ITU-T #7 signaling protocol, and the SS7 protocol. SCCP provides additional routing and management functions for the transfer of messages, other than call set-up, between signaling points. SCCP provides additional functions to Message Transfer Part (MTP), and typically supports Transaction Capabilities Application Part (TCAP).

SIP — Session Initiation Protocol. SIP is an application layer protocol used to establish, modify, and terminate conference and telephony sessions over IP-based networks.

STUN — Simple Transversal of UDP (User Datagram Protocol) Through NATs (Network Address Translators). As defined by *The Internet Society*: A protocol that allows applications to discover the presence and types of NATs and firewalls between them and the public

Internet. It also provides the ability for applications to determine the public Internet Protocol (IP) addresses allocated to them by the NAT. STUN works with many existing NATs, and does not require any special behavior from them. As a result, it allows a wide variety of applications to work through existing NAT infrastructure.

T

Throughput — The actual amount of useful and non-redundant information which is transmitted or processed.

Trace route — The capability to evaluate the hops taken from one end of a link to the other on a TCP/IP network.

U

UDP — User Datagram Protocol. A connectionless protocol with very limited error recovery capabilities.

UNISTIM — Unified Networks IP Stimulus Protocol. The VoIP call control standard used on Nortel systems.

V

VoIP — Voice over IP. Technology used to transmit voice conversations over a data network using the internet protocol.

Index

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