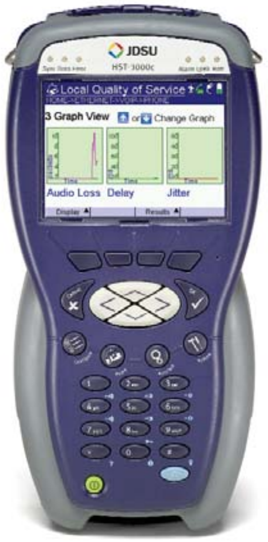
HST-3000

Voice over IP (VoIP) Service Interface Module (SIM)



Key Features

- VoIP phone emulation for service turn-up and troubleshooting
- Supports emulation and monitoring modes
- VoIP voice packet quality assessment and voice quality rating using Mean Opinion Score (MOS) and R-Factor
- Supports Cisco SCCP, SIP, MGCP, H.3223, and Nortel Unistim signaling protocols
- Real-time VoIP and Ethernet statistics
- IP Ping and ICMP/UAP trace route testing
- Supports packet capture and filtering
- AutoAnswer mode for two-ended testing that requires only one technician
- Modular hardware and software architecture that is flexible and easily upgraded to allow testing of multiple services

The JDSU HST-3000 VoIP tester is a versatile field solution for Voice over Internet Protocol (VoIP) service turn-up and troubleshooting. Handheld, rugged, and easy-to-use, the HST-3000 can validate VoIP service connectivity, feature availability, and voice quality. In addition, it provides comprehensive features, including signaling, IP ping, packet statistic and traceroute analysis to identify, diagnose, and sectionalize VoIP network and equipment problems.

VoIP services are widely deployed and to deliver them competitively and cost-effectively, service providers rely upon easy-to-use, multi-function, standards-based field tools that enable technicians to verify service, check voice quality, and troubleshoot problems.

The HST-3000 has advanced, automated test functions, custom scripting, and one-button operation that ensure consistent, accurate, and repeatable test method use, allowing the rapid, efficient, and cost-effective VoIP service directory.

VoIP Service Turn-up

To ensure successful VoIP service turn-up, connectivity to the signaling gateway, feature availability, and call quality must be proven.

The simplest and fastest way to verify connectivity is to place an actual VoIP call. The HST-3000 can emulate an IP phone and supports placing and receiving VoIP calls utilizing Cisco SCCP, SIP, MGCP, H.323, and Nortel Unistim signaling gateways. Calls can be placed to different endpoints to verify translation provisioning—IP phone to IP phone, site to site, IP to TDM network, to a provisioned automated test line, or to another HST-3000—either manned or in AutoAnswer mode.

Both subjective and objective voice quality measurements can be gathered during these connectivity test calls. The HST-3000 provides a packet-based objective measurement of VoIP out of sequence packets call quality by analyzing delay, jitter, and packet loss to generate a good, fair, or poor quality rating based on configurable quality of service (QoS) score thresholds.

Additionally, the HST-3000 uses the patented Telchemy single-ended live call method to provide a real-time assessment of subjective voice quality in terms of both MOS and R-Factor. The valuable data this analysis provides is compared with the data from the objective measurements to quickly verify acceptable VoIP call quality. Call feature provisioning and the availability of supplementary services can also be verified during these calls.

The HST-3000 can also be used to turn-up a VoIP service without the required gateway signaling support, which typically requires end-to-end testing using two HST-3000s. One unit, set to AutoAnswer mode, is connected to the router at the customer premises. The second HST-3000 unit is used to place calls back to the first unit from different endpoints. The first unit answers the call and plays a pre-recorded message. Test measurements are obtained from the pre-recorded message and displayed on the HST-3000.

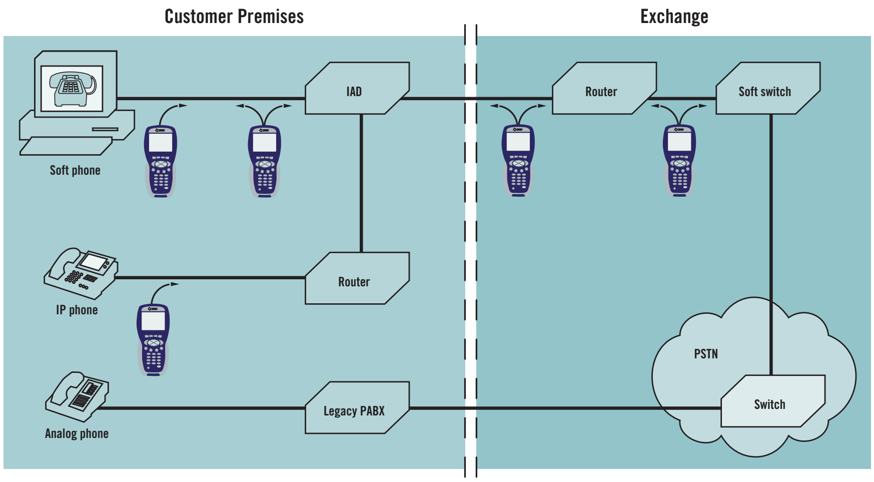


Figure 1 Testing with HST-3000 in a VoIP network

Figure 2 Datacom Configuration Menu

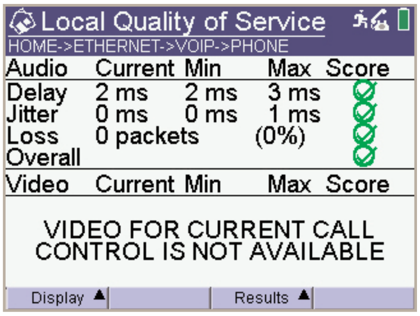


Figure 3 Quality of Service analysis

VoIP Troubleshooting

For unsuccessful calls or poor voice quality, the HST-3000 can be used to identify and sectionalize problems.

Call set-up problems require the ability to troubleshoot the call signaling process. The HST-3000 provides real-time visibility of the entire call set-up process. It displays signaling message decodes and signaling error messages thus allowing quick and easy identification of problems.

Performing IP ping and ICMP or UDP traceroute analysis can isolate path/device connectivity problems and sources of delay or can measure call throughput. Additionally, Ethernet statistics are generated to aid the diagnosis of call quality and identify failed devices or network over utilization.

Reduce Costs, Increase Productivity, and Improve Efficiency

The HST-3000 provides a number of powerful features that can significantly improve the VoIP service turn-up and troubleshooting process, reducing costs and improving productivity and efficiency.

A straightforward graphical user interface (GUI) greatly simplifies the testing process, thus reducing the amount of training needed for comprehensive testing.

In AutoAnswer mode, accomplish two-ended testing across the VoIP network using a single technician. Additionally, one-button automatic testing combined with support for all phases of VoIP service deployment reduces the number of technicians required to turn-up and troubleshoot service, and makes it possible to non-experts to operate tests.

The HST-3000 also offers pre-programmed tests and customized scripts that help ensure that all technicians follow the same procedures, thereby speeding-up service delivery and minimizing installation and testing errors.

Flexible and Rugged Design

The HST-3000 incorporates a rugged, weather-resistant design, and long battery life that are ideally suited for use in the field. Its modularity allows for field upgrades to support new testing requirements. Standard Ethernet, USB, and serial connections offer flexibility to easily download software and offload captured test data for later analysis. Easily configurable, technicians with differing responsibilities can use the HST-3000 to perform a variety of tests. The HST-3000 is easily upgradeable with technologies and advanced options that support the changing needs of service installers.

Figure 4 The architecture of the HST-3000 enables fast, easy field-swapping of a wide variety of test modules

Specifications

Test Ports/Interface Support

10/100 Ethernet (configurable-Half/Full Duplex auto detect), RJ45
ADSL1/2/2+, G.SHDSL 2/4 wire, VDSL1/2 (Modem port 8 pin modular-line on center pins)
USB1.1 Host
RS232 9 pin DIN serial port

Supported Signaling Protocols

H.323 ITU-T H.323 version 3 Fast Connect
H.323 ITU-T H.323 version 3 Full Connect (MSD, CAPSET, OLC exchange)
Skinny Cisco Client Protocol (SCCP)
RTP/RTCP RFC 1889 and 1890
SIP RFS 3621
Nortel Unistim and Secure Unistim
MGCP
Secure RTP for VoIP

Supported Codec Configuration

ITU-T G.711 u-law/A-law (PCM/64 kbt/s)
ITU-T G.723.1 (ACELP/5.3, 6.3 kbt/s)
ITU-T G.726 (ADPCM/16, 24, 32, 40 kbt/s)
ITU-T G.729a (GS-ACELP/8 kbt/s)
ITU-T G.729ab (GS-ACELP/8 kbt/s)
ITU-GSM—FR
ITU-GSM—EFR
User-selectable Silence Suppression, Jitter Buffer, and voice packet size
User-selectable transmit source (Live Voice conversation, tone transmit (200-5 kHz), pre-recorded wave file (up to 2 Mb)

LAN Settings

User-selectable Calling Alias
User-selectable IP address, static, or DHCP
User-selectable subnet mask, gateway, and DNS server
User-selectable or default MAC address
VLAN configurable—IEEE.802.1p/q
Configurable IP TOS

Gatekeeper Settings

User-selectable Static/Auto Discovery, or no gatekeeper direct connect mode
Supports inbound and outbound calls with or without gatekeeper support

Reported Results—VoIP

Full incoming call statistics, including IP address, Alias, Name, RTCP availability/ports, Codec and rate, call signaling support, silence suppression enabled, and call duration
Throughput sent/received in bytes and packets, out of sequence packets
Call progress and signaling error messages
Packet delay (min/max/avg)
Packet jitter (min/max/avg)
Packet loss (min/max/avg)
Encoding, packetization, buffering, and total delay
Voice Quality Rating based on packet metrics thresholds set by user
MOS rating, R-Factor, and Voice Degradation Factors supports packet capture and filtering (save internally in USB Mass Memory Storage)

Reported Results—Ethernet TE

Link status, link speed, link duplex detection
Ethernet statistics: collisions, Tx/RX (bytes, frames, errors dropped)
Ping ICMP and UDP statistics: echos sent/received, Ping delay (min/max/avg/cur), lost count/percentage
Supports IP address or DNS name destination
Traceroute ICMP and UDP statistics: hop count, name lookup, and IP address of hops
Supports IP address and DNS address destination

Physical

Size (H x W x D)	241 x 114 x 70 mm (9.5 x 4.5 x 2.75 in)
Weight (with battery)	1.23 kg (2.7 lb)
Operating temperature	5.5 to 50°C (22 to 122°F)
Storage temperature	–40 to 65.5°C (–40 to 150°F)
Battery life	10 hrs typical usage
Charging time	7 hrs from full discharge to full charge
Operating humidity	10 to 80% relative humidity
Storage humidity	10 to 95% relative humidity
Display	3.8″ diagonal, 1/4 VGA, Color Active Matrix with backlight (readable in direct sunlight)

General

Ruggedness	Survives 91 cm (3 ft) drop to concrete on all sides
Water-resistant	Splashproof (may be used in heavy rain)
Languages	English, German, French, Spanish, Italian, Chinese, Turkish
Keypad	Typical 12-button keyboard

Ordering Information

Base Unit

HST3000-NG	HST-3000 Mainframe without Copper (Color)
HST3000C-NG	HST-3000 Copper Mainframe (Color)

Available SIMS (Modules)

HST3000-CUCE	Copper only SIM, CE Marked
HST3000-AR2A-T1	ADSL2+ T1 (ATU-R, Annex A)
HST3000-AR2A	ADSL1/2/2+ (ATU-R, Annex A)
HST3000-AR2B	ADSL1/2/2+ (ATU-R, Annex B)
HST3000-AR2B-T1	ADSL2+ T1 (ATU-R, Annex B)
HST3000-CAR2A	ADSL1/2/2+ with Copper (ATU-R, Annex A)
HST3000-CAR2A-T1	Copper, ADSL2+ T1 (ATU-R, Annex A)
HST3000-CAR2B	ADSL1/2/2+ with Copper (ATU-R, Annex B)
HST3000-CAR2B-T1	Copper, ADSL2+ T1 (ATU-R, Annex B)
HST3000-CARB	Annex B Copper/ATU-R
HST3000-CARCA	Copper and ATU-R/C Dual Mode, AoPOTS
HST3000-CARCB	Copper and ATU-R/C Dual Mode, AoISDN
HST3000-CARCE	Copper and ATU-R (Annex A), CE Marked
HST3000-WB2	Wide Band 2 (up to 30 MHz) Copper Test
HST3000-VDSL-CNXT	VDSL with Connexant Chipset
HST-3000-VDSL-CNXT-WB2	VDSL and Copper (up to 30 MHz) with Connexant Chipset
HST3000-VDSL-IK	VDSL with Ikanos Chipset
HST-3000-VDSL-IK-WB2	VDSL and Copper (up to 30 MHz) with Ikanos Chipset
HST3000-INF-VDSL	VDSL with Infineon Aware Chipset
HST-3000-INF-VDSL-WB2	VDSL and Copper (up to 30 MHz) with Infineon Aware Chipset
HST3000-ETH	10/100/1000 Ethernet
HST3000-CT1	T1 and Copper
HST3000-DC	Datacom
HST3000-E1	E1
HST3000-E1-DC	E1/Datacom
HST3000-4WLL	4-Wire Local Loop
HST3000-T1	Dual TX/RX Bantam T1 Interface and T1
HST3000-T3	Dual TX/RX Bantam T1 Interface, and Dual RX/Single TX BNC DS3 Interface/and DS3
HST-BRA	ETSI (Euro) ISDN BRA
HST3000-BRI	ISDN BRI
HST3000-CSHCE	G.SHDSL and Copper
HST-GSH	G.SHDSL
HST3000-GSHCE	2-Wire G.SHDSL
HST3000-CSH4	Copper, 4-Wire G.SHDSL (STU-R/C, Annex A/B)
HST3000-BLK	Blank

Software Options

HST3000-BLUETOOTH	Bluetooth Wireless
HST3000S-WEB	Web Browser
HST3000-REMOP	Remote Operation
HST3000-SCRIPT	Scripted Test
HST3000-DSL2	ADSL2 and ADSL2+
HST3000S-IP	Advanced IP Suite—PING and Through Mode Support
HST3000S-IP-Video	IP Video Analysis
HST3000S-VMOS	Video MOS Analysis
HST3000-MSTV	Microsoft IPTV Video Analysis
HST3000-VT100	VT100 Emulation
HST3000S-VOIP	VoIP Software Analysis
HST3000S-H.323	H.323 VoIP Signaling
HST3000S-MGCP	SCCP MGCP VoIP Signaling
HST3000S-MOS	VoIP Mean Opinion Score
HST3000S-SCCP	SCCP VoIP Signaling
HST3000S-SIP	SIP VoIP Signaling
HST3000-UNISTIM	VoIP Signaling Call Controls for UNISTIM
HST3000-OPTETH	Optical Ethernet
HST3000-IPV6	IPv6
HST3000-MPLS	MPLS
HST3000-MSTR	Multiple Streams
HST3000-TCPUDP	TCP/UDP
HST3000-FTP	FTP
HST3000-WBTONES	WB TIMS
HST3000-PCMTIMS	TIMS (PCM)
HST3000-PCMSIG	Signaling (PCM)
HST3000-SPE	Spectral Noise
HST3000-RFL	RFL
HST3000-TDR	TDR
HST3000-PRI	ISDN PRI (NC Standard)
HST3000-ST	Basic Rate ISDN S/T (ANSI)
HST3000-T1DDS	DDS-T1
HST3000-TxIMP	Transmission Impairments
HST3000-FR	Frame Relay
HST3000-PS	Pulse Shape

Test & Measurement Regional Sales

NORTH AMERICA	LATIN AMERICA	ASIA PACIFIC	EMEA	www.jdsu.com/test
TEL: 1 866 228 3762	TEL: +1 954 688 5660	TEL: +852 2892 0990	TEL: +49 7121 86 2222	
FAX: +1 301 353 9216	FAX: +1 954 345 4668	FAX: +852 2892 0770	FAX: +49 7121 86 1222	

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