Abacus™ 50 Ethernet Test System

VOIP, SIP, IMS, RTP, RTCP AND VOICE QUALITY DISTRIBUTED TESTING

The Abacus 50 Ethernet test system simulates VoIP calling functionality by originating and terminating calls in a cost-efficient standalone platform designed for laboratory and distributed testing. Abacus 50 Ethernet troubleshoots and monitors any channel for verification and analysis while generating and terminating calls.

APPLICATIONS

VoIP Convergence

- Measure one-way delay between TDM and VoIP devices
- Verify functionality of media and voice gateways
- Assess voice quality

Network Equipment Manufacturers (chips, IP-PBX, gateway, MSs and SSs)

- Characterize system before trial
- Validate system scalability

Service Providers (NSPs, SPs, ITSPs, and Enterprises)

- Facilitate vendor selection
- Enable accurate capacity/network planning and deployment analysis
- Provide end-to-end service assurance testing

BENEFITS

- Simplify the testing of converged IP Telephony and PSTN networks and services with functional and performance testing for SIP, SIP-T, SIP-I, Skinny, H.323, MGCP/NCS, and H.248/Megaco
- Enable service providers and enterprises to reduce time to market of services, while assuring they meet quality requirements as perceived by users
- Achieve overall cost savings by offering full flexibility in convergence testing with synchronized Abacus 50 Ethernet, Analog, T1/E1 in addition to Abacus 5000 VoIP, TDM, Analog measurements, all using the same user interface
- Get affordable, low-cost scalability for laboratory or distributed test environments that require few thousands simulated VoIP callers per test stand

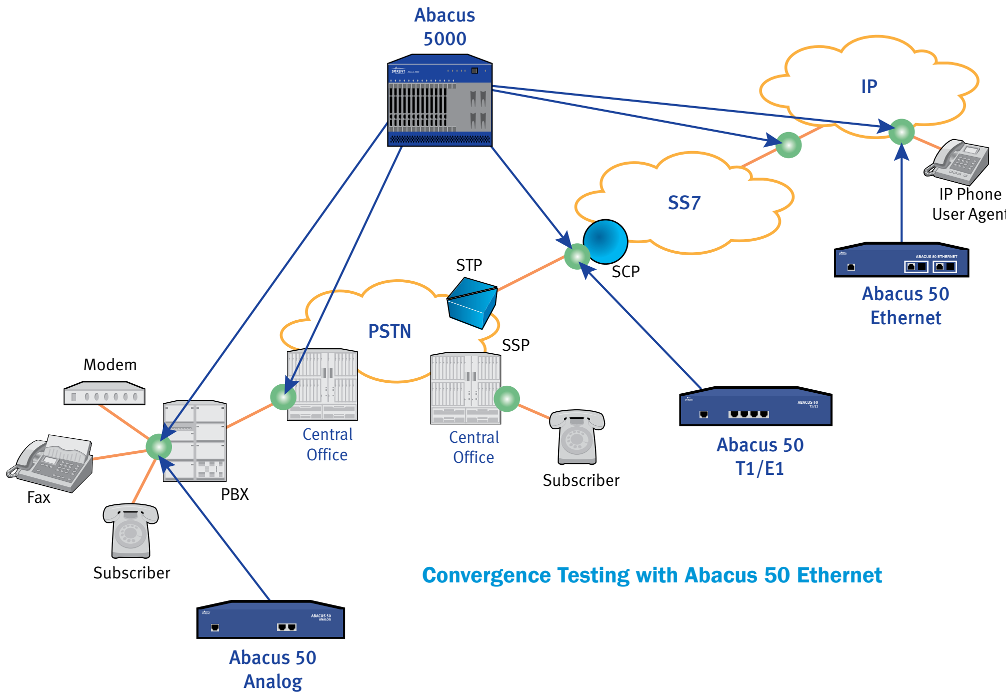
OVERVIEW

The Abacus 50 Ethernet Test System is available in 2 options:

- Abacus 50 GigE
- Abacus 50 10/100 Eth

Abacus 50 GigE supports 2-port 10/100/1000Base-T, Fiber Gigabit Ethernet.

Abacus 50 10/100 Eth supports 2-port 10/100Base-T Ethernet and can easily be upgraded to Gigabit Ethernet.



Convergence Testing with Abacus 50 Ethernet

Abacus 50 Ethernet can be controlled by the same user interface, as the Abacus 50, Abacus 100 or Abacus 5000, for inexpensive testing of low-density VoIP convergence equipment.

Deploying a remote testing solution requires a stable test tool that generates thousands of calls. With the distributed testing option, multiple systems can be viewed as one system for simplified management of multiple Abacus 50 Ethernet test systems (along with Abacus 5000, Abacus 100, Abacus 50 Analog, and Abacus 50 T1/E1 test systems).

The Abacus 50 Ethernet test system simulates multiple IP telephones and/or gateways, generating the call signaling, and delivering the signaling and/or media traffic to a system under test.

The Abacus 50 Ethernet test system tests VoIP gateways, VoIP PBXs, gatekeepers, proxy servers, media gateway controllers, softswitches, and other internetworking gateways and PSTN equipment.

ABACUS 50 ETHERNET TEST SYSTEM FEATURES

Voice Quality Testing

- Perform voice quality measurements on 64 channels using MOS, PSQM, and PSQM+
- Perform voice quality measurements on 32 channels using PESQ, MOS-LQO, R-Factor (P.834), and J-MOS
- Perform voice quality measurements on 16 channels using PESQ-WB
- Abacus 50 Ethernet performs voice quality measurements on thousands of channels using E-Model – R-Factor (G.107)

Quality of Service (QoS) Testing

- Perform QoS validation on 32 channels using the Scripting for Voice Pattern Matching

Voice Security Testing

- Test SIP IMS Security with IPSec and AKA
- Test SIP Security with Secure RTP and TLS

VoIP Signaling and Media Traffic Generation

- Generate VoIP calls for SIP, Skinny Client Control Protocol (SCCP), H.323, MGCP, NCS, H.248/Megaco, SIP-T, SIP-I, T.38 fax over IP, T.30 Fax, 3PCC, and H.235-GRQ
- Generate and/or terminate signaling-only calls without voice
- Generate or terminate media calls (RTP and signaling) with path confirmation
- Generate call setups to media gateway controllers, gateways, or softswitches

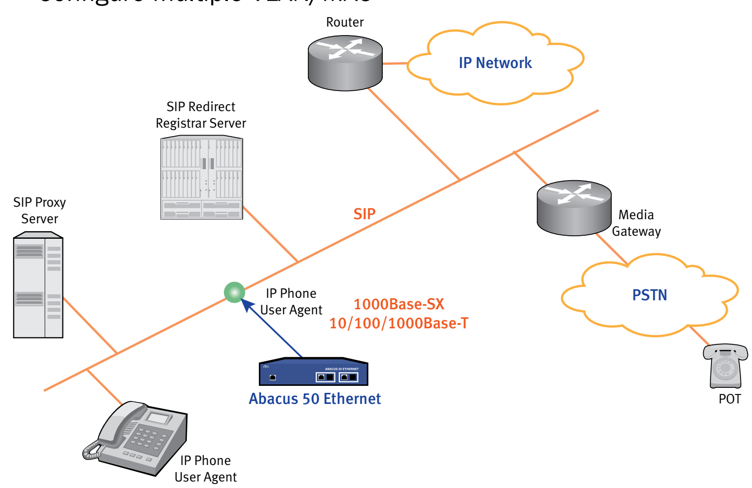
- Originate and terminate calls with 2-port dual media Gigabit Ethernet
- Load Profiling (Saw Tooth, Rectangle, Trapezoid and Poisson)
- Network Topology Diagram with predefined diagrams

Troubleshooting and Diagnostics Capabilities

- Verify a speech path is established and retained for the duration of the call
- Measure delays, call setup time, lost packets, out of order packets, jitter, call attempts per second, and call completions
- Generate detailed call error reports with sequences or messages that failed
- Continuously gather and present results in tables and graphs automatically
- Track signaling history only on the channels where errors occurred with the event AutoTrack, providing detailed failure information around the time of the event
- Monitor any channel for verification or analysis
- Display messages in full decode
- Capture VoIP data during call generation and without call generation
- View time between messages
- Graphical display of Measurements-over-Time
- Measure one-way delay measurements
- Support transport protocols
  - RTP/RTCP
  - TCP/UDP
- Send/receive tones, speech, using G.711 (μ/A-Law), G.723.1, G.726, G.726A, G.729AB
- Support Mobile-NB GSM-AMR, EVRC, AMR-WB (G.722.2), iLBC, and GSM EFR
- Support video using H.263 and H.264
- Support VAD – Voice Activity Detection as a CODEC parameter
- Support Cisco VAD
- Built-in protocol decoding and display
- Provide configurable protocol development
- Send and receive tones, speech, and video
- Detect and forward DTMF tones
- Support flexible call sequences
- Support TOS/Diffserv for RTP
- Support toggle
- Measure echo
- G.711 encoded T.30 Fax over RTP
- Support up to 512 actions per script

SUPPORT FOR SIP

- Support thousands of simultaneous registered SIP users
- Generate and/or terminate signaling-only calls without voice
- Support IPv6/Unicast addressing and IPv6 traffic class on 2-port
- Support T.38 fax over SIP and call tracer (ladder diagram) for T.38
- Configurable SIP call flow and message generation
- Configurable SIP ports
- Configurable SDP
- Configurable RTP ports
- Filter protocol messages displayed by message type or message headers for SIP signaling
- Support TOS/Diffserv (DSCP) for SIP signaling
- Support 3PCC – 3rd Party Call Control RFC 3725, and automatic detection with SIP protocol development
- Support VLAN tagging according to the IEEE standard 802.1Q
- Configure multiple VLAN/MAC



SIP Testing with Abacus 50 Ethernet

- Support stacked VLAN QinQ
- Support multi-proxy (2 SIP proxies) functionality as described in RFC 3665 Section 3.3
- Support multi-proxy server through SIP scripting
- Support video and voice streams in the same call within the SIP session
- Test VoIP Security with Secure RTP and TLS
- Support configuration of RTP H.264 video
- SIP Scripting
- Call Tracer (Ladder diagram)
- SIP Message Editor
- Support for 3GPP (RFC 3891), RFC 3312, RFC 3608 6.1 (SIP “Replace” Header functionality: INV w/Replaces)
- 3GPP Call Flow and Registration API
- Verify IP addresses of incoming SIP calls and hunt group for SIP
- Trunk events for RFC 2833
- PhoneBook import/export
- TCL API for SIP PhoneBook
- VLAN Tags for Turbo RTP
- RTCP on Turbo RTP
- Send phone numbers by DTMF without use of DSP for at least 1000 SIP endpoints
- Gradual increase RAPS over time
- DHCP for IPv4
- DHCP-PD (IPv6 support)
- SIP Proxy emulation registrar and call routing
- RTP Replay
- IMS Security (IPSec and AKA)
- SIP Signaling Compression (SIG Comp)
- SIP Supplementary Feature Testing
- SIP Trunking
- Mobile SIP UE
- SIP Loopback

SUPPORT FOR SIP-T and SIP-I

- Advanced testing for both trunking and signaling gateways with SIP-T (SIP for Telephones)
- SIP-I (SIP with encapsulated ISUP)

SUPPORT FOR SKINNY

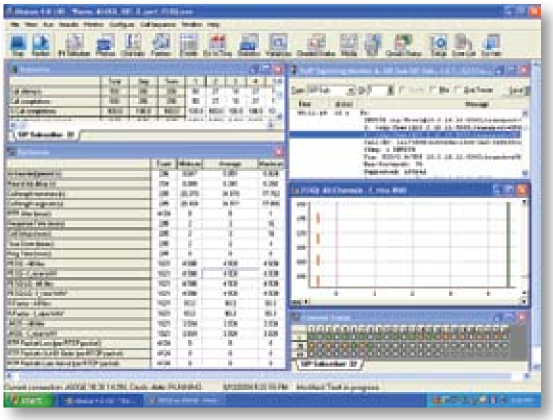
- Emulate Skinny client end-points and stress Cisco Call Manager (CCM)
- Emulate Cisco Call Manager (CCM)
- Audio to Conference Server
- Multiple VLAN/MAC configuration
- TOS/Diffserv

H.323 SUPPORT

- Support thousands of H.323 end-point registrations
- Send bearer capability information within “speech” using H.323
- Support H.323 IRR call status report
- T.38 fax over H.323
- Configurable Gatekeeper ReQuest (GRQ) - H.235 VoIP gateway user credential encryption
- Functional and load testing specific to GRQ messages and H.235 signaling encryption
- Multiple VLAN/MAC configuration
- H.323/Q.931
- Call Tracer (Ladder diagram)
- Multiple H.323 calls over single TCP connection
- Multiple H.323 EP’s per Unique/Single IP address

H.248/MEGACO SUPPORT

- All-in-one media gateway emulation
- H.248/Megaco call routing/switching
- H.248.1/Megaco versions 1 and 2 over UDP
- MSF H.248/Megaco
- Binary H.248/Megaco
- T.38 fax over H.248/Megaco
- T.38 fax for Trunking Gateway
- Test VoIP Security with Secure RTP



Voice Quality Testing with Abacus 50 Ethernet

- Call Tracer (Ladder diagram)
- Multiple VLAN/MAC configuration
- ETSI H.248/Megaco
- ServiceChange METHOD

MGCP SUPPORT

- All in one media gateway emulation
- NCS
- MGCP 1.0
- MGCP fully qualified domain name
- MGCP call routing/switching
- T.38 fax over MGCP
- T.38 fax for Trunking Gateway
- Multiple VLAN/MAC configuration
- Test VoIP Security with Secure RTP
- Call Tracer (Ladder diagram)
- Multiple VLAN/MAC configuration
- Message Fragmentation and 521 redirect
- Hook-Flash

TEST METHODOLOGY SPECIFICATIONS

Call Generation

- Tones
- Speech and video
- Path confirmation
- Fax

Tones (G.711 only)

- Send any two frequencies with an accuracy of ±0.05% or ±0.5 Hz
- Send noise or silence
- Send with a resolution of 8 ms and an accuracy of ±20 ms
- Detect any two frequencies with a minimum difference of 80 Hz for no noise
- Detect energy or silence
- Detect signals with a minimum duration of 40 ms at various thresholds, with an accuracy of ±20 ms

Path Confirmation

- 3-tone: use series of three single frequencies
- Physical: use series of dual frequencies to identify unique address of channel
- Resilient: exchange tones with precise Voice Activation Factor (VAF), and measure disturbances in the speech path

Measuring Voice Quality

- PSQM, PSQM+
- PESQ, PESQ-WB
- PSQM to MOS correlation
- MOS-LQO, R-Factor (P.834), and J-MOS calculations from PESQ measure-ments
- E-Model – R-Factor (G.107)

Call Routing/Switching

- Routes and switches H.248/Megaco, and MGCP calls
- Detects off-hook on the originating side
- Collection of digits dialed by the originating subscriber
- Translates the digits
- Terminates subscriber's user address
- Alerts the originating and terminating subscriber of the call
- Instructs the originating and terminating side to connect the call
- Releases the call on completion

Protocol Analyzers

- Contain trace and decode signaling and state machine
- Save traces to text files
- Open multiple trace windows (two monitoring channels each for IP Telephony, line, and data links)
- Access analyzer from user interface
- Capture VoIP data in PCAP format during call generation and without call generation

Speech and Video

- Send any WAV files
- Send video files

Scripts

- Attempts
- Completions

Fax

- T.38
- T.30

Making and Receiving Calls

- Audio monitor: listen to any 2 channels from the controlling PC

VoIP Packet Measurements

- RTP packet loss
- RTP packets out of order
- RTP packets extended delay
- RTP jitter
- SIP response time
- RRQ response time
- Call setup
- Call tear down
- Ring time
- Packets received
- Packets transmitted
- SIP registration response time
- SIP registration success time
- Statistical SIP message count

VoIP Delay Measurements

- Dial tone
- Call setup
- Media path round trip
- One-way delay

PAYLOAD MEDIA SPECIFICATIONS

Voice CODECs

- Encoding and decoding
  - G.711 μ-law and A-law, G.726, G.726 ITU-T, G.726A, G.723.1, G.729A/B
  - G.726 ITU-T, G.726A at 16 Kbps, 24 Kbps, 32 Kbps and 40 Kbps
  - Wireless NB GSM-AMR and EVRC
  - AMR-WB (G.722, G.722.1, G.722.2)
  - EVRC-B
  - iLBC
  - GSM EFR, FR
- Encoding (only)
  - Wireless NB 1/2 rate EVRC

Video CODECs

- H.263 and H.264 encoding for SIP

Echo Measurements

- Echo cancellation on/off
- Echo delay
- Echo Return Loss (ERL)
- Echo Return Loss Enhancement (ERLE) measurement
- Talk Echo Loudness Rating (TELR) measurements
- Support echo measurements on 2 channels

PROTOCOL SPECIFICATIONS

IP Telephony

- SIP IETF RFC 3261, RFC 3312, 3GPP (RFC 3891), RFC 3608 6.1
- Skinny Client Control Protocol (SCCP) version 4.1
- ITU-T H.323 versions 2.0 and 4.0
- Configurable GRQ (Gatekeeper ReQuest) - H.235 VoIP gateway message encryption
- MGCP IETF RFC 3435
- H.248.1/Megaco v1 IETF RFC 3525 and H.248.1 v2 Megaco in the IETF and as per Recommendation H.248 (now H.248.1) in the ITU-T
- SIP-T RFC 3372
- RTP/RTCP RFC 3550, RFC 3551 and RFC 2833
- SDP RFC 2327 and RFC 3108
- T.38 Fax over SIP, H.323 and H.248/ Megaco, MGCP

PHYSICAL SPECIFICATIONS

Components

- Standalone 1U high 19" rack mountable unit with included brackets

Dimensions

- Height: 46.4 mm (1.825") with user installed feet; 43.2 mm (1.7") excluding feet
- Width: 24 cm (9.5")
- Depth: 20 cm (8")

Weight

- 1.8 Kg (4 Lbs.)

Environment

- Operating temperature range: 0 – 40°C at 10% – 70% non-condensing humidity
- Altitude: –100 to +4000 m (–328 to +13,123 feet)
- CE marked

LEDs

- 2 LEDs per port indicate SUT activity (ACT) and link (LNK) status at the active connector
- Status LED indicates Abacus 50 Ethernet Test System health
- 2 LEDs on LAN connector indicate LAN activity (ACT) and link (LNK) status

Interfaces for Abacus 50 GigE

- 2-port dual media Gigabit Ethernet with 1000Base-SX transceivers included

Interfaces for Abacus 50 10/100 Eth

- 2-port Ethernet with one 10/100/1000 Base-T transceiver per port included
- Upgradeable to 2-port dual media Gigabit Ethernet with one 10/100/1000Base-T transceiver and one 1000Base-SX transceiver per port

Front panel connections for Abacus 50 GigE

- Front panel with two RJ-45 Ethernet 10/100/1000Base-T connectors, two SFP fiber-optic connectors (1000Base-SX transceiver included, 1000Base-LX transceiver is optional) and one RJ-45 10/100Base-T Ethernet console LAN connector

Front panel connections for Abacus 50 10/100 Eth

- Front panel with two RJ-45 Ethernet 10/100Base-T connectors and one RJ-45 10/100Base-T Ethernet console LAN connector
- SFP fiber-optic connectors 1000Base-SX transceiver or 1000Base-LX transceiver are optional (P/Ns ICG-3001SX, ICG-3001LX)

Back panel connections for Abacus 50 Ethernet Test System

- Back panel with one -48 VDC power connector, grounding post, DB15 connector for future GPS/CDMA time synchronization (currently supported through Ethernet), and one DB9 serial port connector

ELECTRICAL

- Power supplied through external -48 VDC desktop power supply with locking connector or external -48 VDC source
- 90 to 264 VAC (47 to 63 Hz) or -36 to -72 VDC
- Power draw: Maximum of 50W, 35W typical
- Power switch on back panel with fuse

ORDERING INFORMATION

Abacus 50 GigE

- Abacus 50 – GigE 2-port dual media (P/N A-50-006)
  - 1000Base-LX GIG Ethernet SFP transceiver, SM 1310NM, LC CONN (P/N ICG-3001LX)

Abacus 50 10/100 Eth

- Abacus 50 10/100 Eth (P/N A-50-008)
  - Upgrade Abacus 50 10/100 Eth to 10/100/1000Base-T, Gigabit (P/N SWF-3740)
  - 1000Base-LX GIG Ethernet SFP transceiver, SM 1310NM, LC CONN (P/N ICG-3001LX)
  - 1000Base-SX GIG Ethernet SFP transceiver, MM 850NM, LC CONN (P/N ICG-3001SX)

Multi-System/Distributed Firmware Option

Required when using more than one Abacus 50, or with any combination of Abacus 5000 or Abacus 100. Abacus 5000 must also have distributed option enabled (P/N SWF-3210). Distributed option is included with Abacus 100.

- Enable Abacus system for distributed testing (P/N SWF-3510)

Firmware Options

The firmware options listed below are available for both Abacus 50 GigE and Abacus 50 10/100 Eth. The SWF-36xx are required for Abacus 50 GigE. The SWF-37xx are required for Abacus 50 10/100 Eth.

- SIP for VoIP call control (P/Ns SWF-3601, SWF-3701)
- MGCP/NCS for VoIP call control (P/Ns SWF-3602, SWF-3702)

- H.248/Megaco on UDP (P/Ns SWF-3603, SWF-3703)
- H.323 for VoIP call control (P/Ns SWF-3604, SWF-3704)
- SIP-T, SIP-I call control (P/Ns SWF-3605, SWF-3705)
- PSQM, PSQM+ (P/Ns SWF-3609, SWF-3709)
- PESQ (P/Ns SWF-3610, SWF-3710)
- Video Encoding (H.263) (P/Ns SWF-3614, SWF-3714)
- RTP Video (H.264) (P/Ns SWF-3617, SWF-3717)
- T.38 Fax over IP (P/Ns SWF-3618, SWF-3718)
- IPv6 for SIP and RTP (P/Ns SWF-3619, SWF-3719)
- Skinny Client Control Protocol (P/Ns SWF-3620, SWF-3720)
- T.30 Fax up to V.17 (P/Ns SWF-3622, SWF-3722)
- Echo measurements (P/Ns SWF-3626, SWF-3726)
- Call routing/switching (P/Ns SWF-3628, SWF-3728)
- Stacked VLAN, QinQ (P/Ns SWF-3629, SWF-3729)
- SRTP (P/Ns SWF-3632, SWF-3732)
- TLS (P/Ns SWF-3633, SWF-3733)
- E-Model (P/Ns SWF-3634, SWF-3734)
- SIP Scripting (P/Ns SWF-3635, SWF-3735)
- Scripting for Voice Pattern Matching (P/Ns SWF-3639, SWF-3739)
- RTCP on Turbo RTP (P/Ns SWF-3643, SWF-3743)
- DHCP (P/Ns SWF-3648, SWF-3748)
- PESQ-WB (P/Ns SWF-3649, SWF-3749)
- SIP IMS Security - IPSec and AKA (P/Ns SWF-3652, SWF-3752)
- AMR-WB (G.722.x) Encoding and Decoding (P/Ns SWF-3653, SWF-3753)
- EVRC-B Encoding and Decoding (P/Ns SWF-3654 , SFW-3754)
- SIP Proxy Emulation (P/Ns SWF-3655, SWF-3755)
- SIP Signaling Compression (SigComp) (P/Ns SWF-3656, SWF-3756)
- iLBC Encoding and Decoding (P/Ns SWF-3659, SWF-3759)
- GMS EFR, FR Encoding and Decoding (P/Ns SWF-3662 SWF-3762)
- RTP Replay (P/Ns SWF-3663, SWF-3763)
- G.723, G.726 Encoding and Decoding (P/Ns SWF-3664, SWF-3764)
- Mobile NB Encoders and Decoders GSM-AMR, EVRC (P/Ns SWF-3665, SWF-3765)

FOR MORE INFORMATION

Visit Spirent Communications' Web site at www.spirent.com/go/voice where you can learn about Spirent IP Telephony test systems and services, download product literature, white papers and test methodologies. Contact your local sales representative for details.

SPIRENT GLOBAL SERVICES

Spirent Global Services provides a variety of professional services, support services and education services — all focused on helping customers meet their complex testing and service assurance requirements. For more information, visit the Global Services Web site at www.spirent.com/gs or contact your Spirent sales representative.

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