SPIRENT ABACUS

SIP FOR VOIP AND IMS

ABACUS 5000—IP TELEPHONY SIGNALING AND TRAFFIC GENERATOR

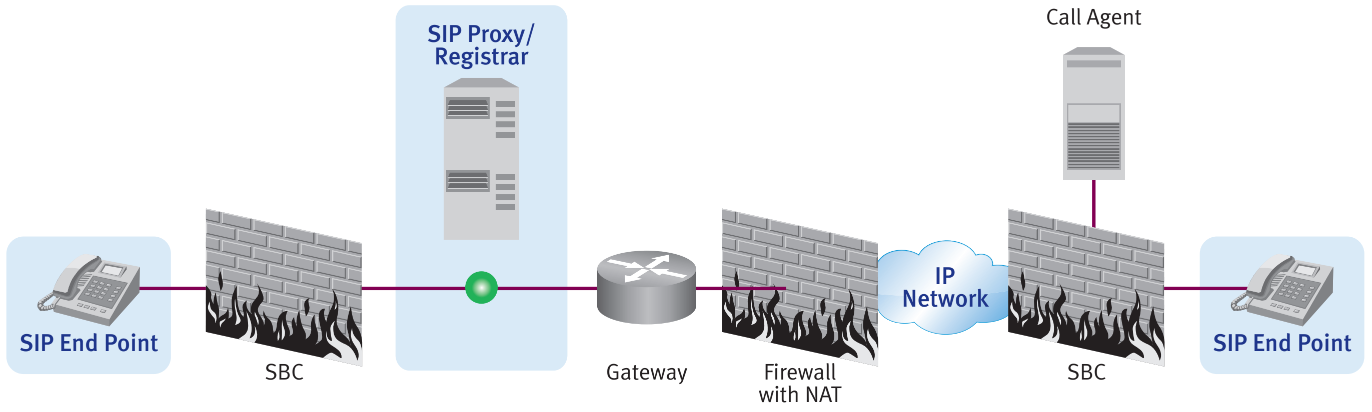
The convergence of PSTN, VoIP over Fixed and Wireless Broadband, and Mobile networks to an all IP infrastructure enables service providers to roll out new services quicker and with greater cost efficiency. Industry experts have chosen the SIP protocol for device intercommunication between users and server network elements to deliver voice and video services.

APPLICATIONS

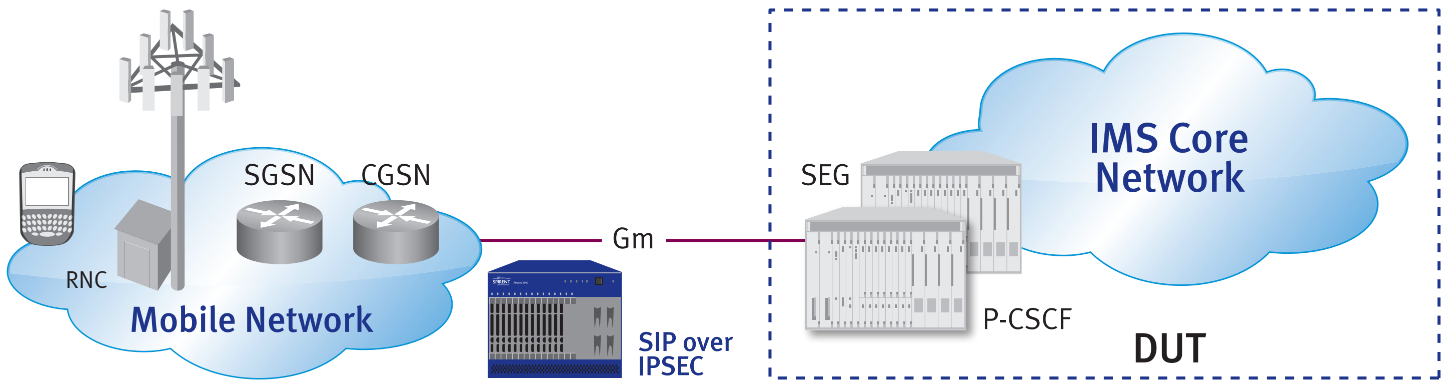
- Assess SIP Signaling Delays
- Assess voice KPI (Key Performance Index)
- Protocol analysis and troubleshoot
- Audio monitoring and subjective voice assessment
- VoIP and IMS Security and Authentication
- Busy Hour Call Attempt simulation
- Isolate SIP network elements
- Distributed live network testing
- Real-world SIP UA emulation

The delivery of next-generation voice service must have the same quality as today's traditional land line. Legacy land line voice services have been in service for more than 100 years. With the next generation network, customers accustomed to the high quality of service provided by land line services will expect the same quality of voice service regardless of the network.

Spirent Abacus™ SIP stack-based solution, based on RFC 3261 and 3GPP IMS specifications, enables Spirent customers to validate key control plane performance of SIP network elements such as Proxy, Registrar and Redirect servers, P-CSCF and IMS security Gateways, Session Border Control and VoIP Firewall. Whether existing in a controlled lab environment, or a pilot or live operational network, Abacus SIP applications measure and validate key control plane signaling delays to ensure high user quality of service by emulating thousands of SIP UAs simulating calls during busy hour. Combined with Abacus TDM and Analog interfaces, Abacus SIP with RTP/RTCP can validate the KPI (Key Performance Index) of voice media elements like media gateways, IP PBX and Media Servers (IVR and Voicemail). All real-time KPI and delay measurement results will enable service providers and NEMs to quickly address any issues and reduce network downtime.



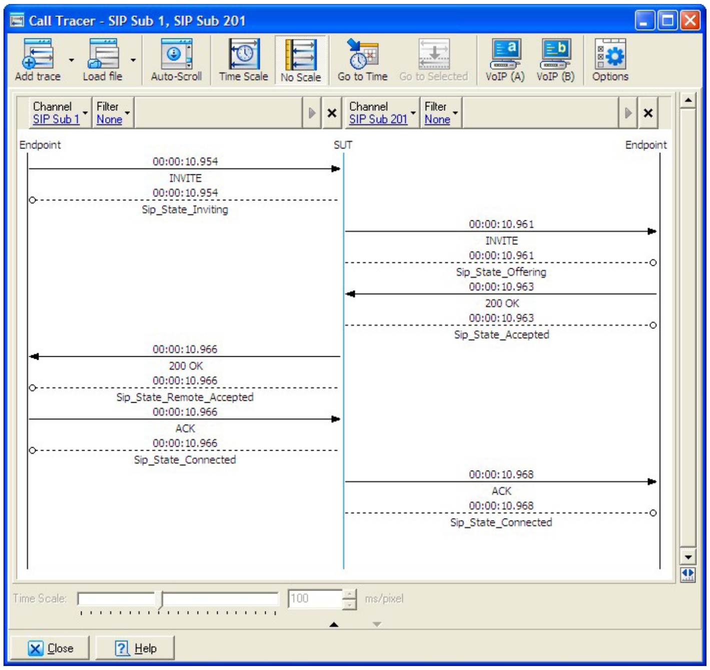
ABACUS 5000—IP TELEPHONY SIGNALING AND TRAFFIC GENERATOR



FEATURES & BENEFITS

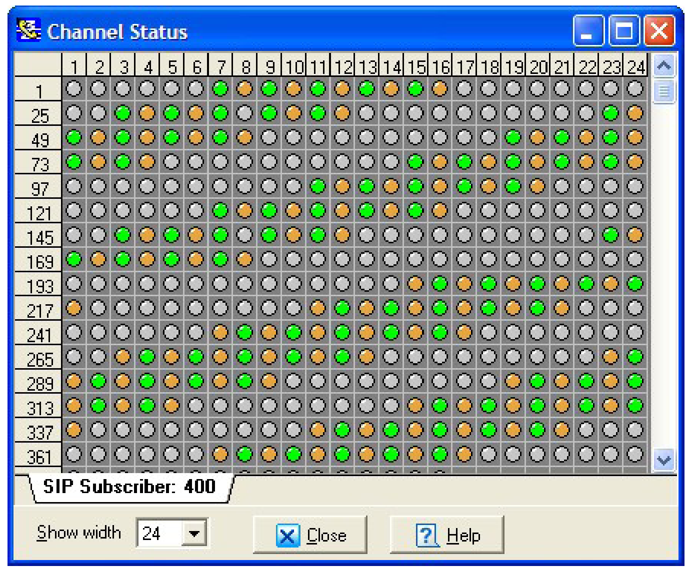
- Fully compliant SIP UE Client/Server Stack—Spirent Abacus SIP supports the call flows of RFC 3261 with an internal pre-built state machine without the challenges of user-defined call flows. The internal state machine will automatically track stateful calls for managing each SIP transactions. Abacus SIP stack can simulate thousands of SIP UEs per port and generate thousands of SIP transactions per seconds. The SIP Stack allows for greater interoperability with many SIP Network Servers. Supports natively MD5 authentication, SIP Registrations/Re-Registrations and SIP Message retransmission timers (T1 and T2).
- SIP Security—Supports IMS IPSec and AKA authentication for stress testing P-CSCF at Gm interface. Abacus SIP also supports SIP over TLS for simulating secured UE to UE communication. Media encryption with SRTP extends security testing.
- IMS SIP UE Emulation—Spirent Abacus SIP supports SIP Mobile UE call flows to simulate thousands of Mobile UEs calling into the IMS network over Gm interface using IPSec protocol for security signaling traffic. Abacus SIP supports IMS aware SIP UE per 3GPP TS 33 203 for stress testing IMS Servers with AKA Digest Authentication and SigComp.
- SIP Protocol Development and Scripting—Protocol development UI provides simple graphical interface for simple SIP syntax modification from changing short-to-long form text, adding proprietary headers and configuring 1xx, 2xx, 3xx, 4xx, 5xx and 6xx responses. For more complex call flows and dynamic SIP parameters settings, the Abacus SIP stack can be modified on the fly with the powerful SIP Scripting feature that allows user to create SIP Supplementary service call flows and to create new industry standard headers.
- SIP Supplementary Feature Testing—Enables users with much greater message and call flow flexibility for SIP via an easy to use graphical editor. Provides configurable SIP to Support Supplementary Feature Testing per RFC 5359 (SIP Services Examples) with the enhanced SIP Message Editor. The SIP Message editor is a SIP Call Flow builder based on “user actions” for user realism emulations.
- SIP Proxy/Registrar Emulation—The SIP Proxy/Registrar emulation along with SIP UE emulation allows complete isolation of SBC, firewall, and QoS-enabled VoIP routers. Broadband VoIP gateways can also be tested in isolation when combined with Abacus PSTN interfaces.
- RTP/RTCP—Spirent Abacus SIP combines with state-of-the-art DSP-based voice test platform to simulate real-world telephone calls. The RTP protocol development tools provide flexibility to customize RTP media profile parameters to cover a broad range of media “voice” testing including packetization, VAD and payload types. By deploying DSP technologies in Abacus call generation subsystems, Abacus vocoders can encode/decode on the fly G.7xx, GSM-AMR NB/WB, GSM-EFR, EVRC/EVRC-B and iLBC codecs for simulation real audio conversations.

- Call Tracer and Ethernet Wire Capture—Abacus applications have enhanced debug capabilities. They assist in debugging SIP signaling issues by capturing and displaying in real time SIP call flows in ladder diagrams for easy to follow SIP signaling transactions. Ethernet captures can be saved in PCAP format for additional lower layer debugging.



- Enhanced SIP Phonebooks—Abacus UI provides the most flexible SIP URI configurations allowing Abacus users to configure thousands of SIP URI in seconds. The SIP phonebook has provisions to configure static or dynamic (DHCP) IPv4/IPv6 address, VLAN and VLAN QinQ, and DiffServ values.
- Real-Time SIP Signaling Statistics—All real time measurements reported to Abacus UI are can be used to assess DUT performance over time. Many telephony related measurements are provided including Call Setup Delay, SIP Response Time, Post Dial Delay, Call Teardown Delay, Calls per second, SIP Registrations per second and Call Duration.

- Media Voice Quality—Active QoS measurements are calculated using DSP by comparing the original and degraded voice files to provide PSQM, PSQM+, PESQ and PESQ-WB scores to accurately evaluate the quality of the media compared to the source. Passive E-Model R-Factor calculations are calculated from the input RTP streams in real time for assessing QoS of voice media with respect to network performance (packet loss, jitter, delays, out of sequence).
- SIP Loopback—Abacus emulates the 'source' peer in the media loopback communications. Abacus simulates both, the call originating and terminating positions. When loopback is set, then Abacus as the 'source', sends and receives the media packets in the negotiated format.



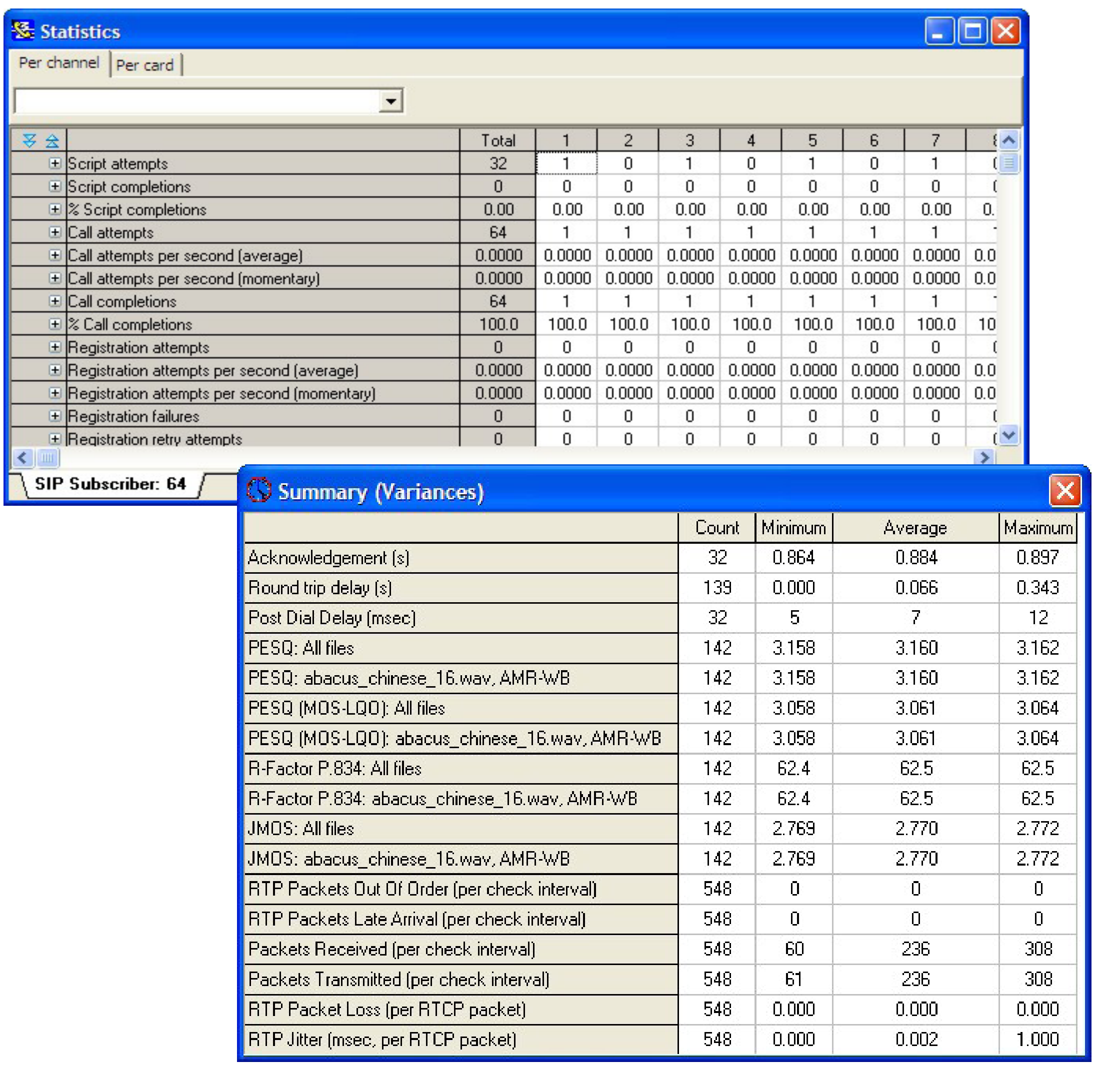
TECHNICAL SPECIFICATIONS

SIP

- RFC 2246 The TLS Protocol Version 1.0
- RFC 2327 SDP: Session Description Protocol
- RFC 2617 HTTP Authentication: Basic and Digest Access Authentication
- RFC 2976 The SIP INFO Method
- RFC 3087 Control of Service Context using SIP Request-URI
- RFC 3261 SIP: Session Initiation Protocol,
- RFC 3262 Reliability of Provisional Responses in Session Initiation Protocol (SIP)
- RFC 3265 Session Initiation Protocol (SIP)—Specific Event Notification
- RFC 3310 Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)
- RFC 3311 The Session Initiation Protocol (SIP) UPDATE Method
- RFC 3312 Integration of Resource Management and Session Initiation Protocol (SIP)
- RFC 3320 Signaling Compression (SigComp)
- RFC 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3325 Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
- RFC 3326 The Reason Header Field for the Session Initiation Protocol (SIP)
- RFC 3327 Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts
- RFC 3329 RFC3329—Security Mechanism Agreement for the Session Initiation
- RFC 3372 Session Initiation Protocol for Telephones (SIP-T): Context and Architectures
- RFC 3455 Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd Generation Partnership Project (3GPP)
- RFC 3485 The Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Static Dictionary for Signaling Compression (SigComp)
- RFC 3486 Compressing the Session Initiation Protocol (SIP)
- RFC 3515 The Session Initiation Protocol (SIP) Refer Method
- RFC 3608 Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration
- RFC 3680 A Session Initiation Protocol (SIP) Event Package for Registrations
- 3GPP TS 33.203 3G security; Access security for IP-based services
- 3GPP TS.24.229 Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)

AUDIO AND VIDEO MEDIA

- RFC 2190 RTP Payload Format for H.263 Video Streams
- RFC 2327 SDP: Session Description Protocol
- RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3550 RTP: A Transport Protocol for Real-Time Applications
- RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control
- RFC 3558 RTP Payload Format for Enhanced Variable Rate Codecs (EVRC) and Selectable Mode Vocoders (SMV)
- RFC 3711 The Secure Real-time Transport Protocol (SRTP)
- RTF 3984 RTP Payload Format for H.264 Video
- G.711 Pulse Code Modulation (PCM) of Voice Frequencies
- G.722.2 Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)
- G.726 40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)
- G.723.1 Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s
- G.729 Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)
- G.729 Annex A: Reduced Complexity 8 kbit/s CS-ACELP Speech Codec
- G.729 Annex B: A silence compression scheme for G.729 optimized for terminals conforming to Recommendation V.70
- T.30 Procedures for Documenting Facsimile Transmission in the General Switched Telephone Network
- T.38 Procedures for Real-time Group 3 Facsimile Communication over IP Networks
- H.263 Video coding for low bit rate communication
- H.264 Advanced Video Coding for Generic Audiovisual Services
- RFC 3267 Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs
- 3GPP TS 26.073 AMR Speech Codec
- 3GPP2 C.S0014-0 v1.0—Enhanced Variable Rate Codec (EVRC)



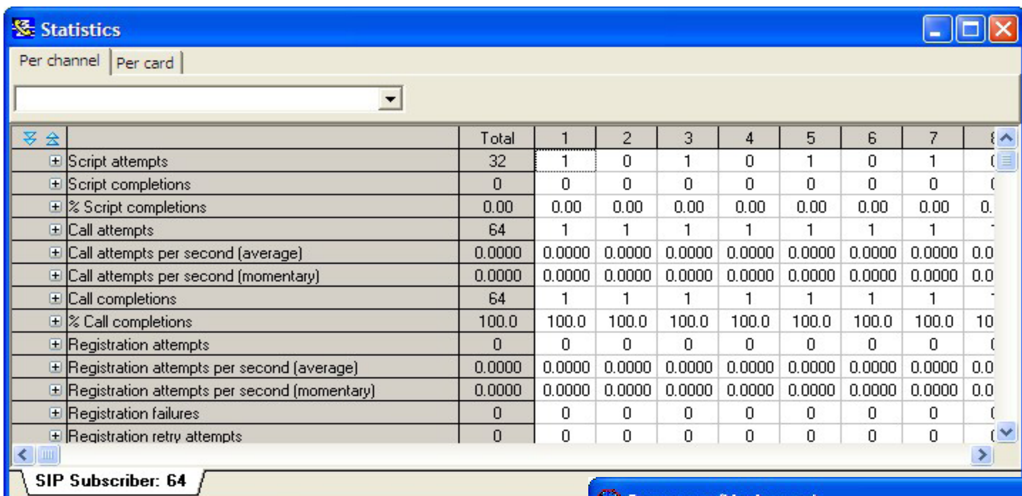
	Total	1	2	3	4	5	6	7
Script attempts	32	1	0	1	0	1	0	1
Script completions	0	0	0	0	0	0	0	0
% Script completions	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Call attempts	64	1	1	1	1	1	1	1
Call attempts per second (average)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Call attempts per second (momentary)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Call completions	64	1	1	1	1	1	1	1
% Call completions	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0
Registration attempts	0	0	0	0	0	0	0	0
Registration attempts per second (average)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Registration attempts per second (momentary)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Registration failures	0	0	0	0	0	0	0	0
Registration retry attempts	0	0	0	0	0	0	0	0

	Count	Minimum	Average	Maximum
Acknowledgement (s)	32	0.864	0.884	0.897
Round trip delay (s)	139	0.000	0.066	0.343
Post Dial Delay (msec)	32	5	7	12
PESQ: All files	142	3.158	3.160	3.162
PESQ: abacus_chinese_16.wav, AMR-WB	142	3.158	3.160	3.162
PESQ (MOS-LQO): All files	142	3.058	3.061	3.064
PESQ (MOS-LQO): abacus_chinese_16.wav, AMR-WB	142	3.058	3.061	3.064
R-Factor P.834: All files	142	62.4	62.5	62.5
R-Factor P.834: abacus_chinese_16.wav, AMR-WB	142	62.4	62.5	62.5
JMOS: All files	142	2.769	2.770	2.772
JMOS: abacus_chinese_16.wav, AMR-WB	142	2.769	2.770	2.772
RTP Packets Out Of Order (per check interval)	548	0	0	0
RTP Packets Late Arrival (per check interval)	548	0	0	0
Packets Received (per check interval)	548	60	236	308
Packets Transmitted (per check interval)	548	61	236	308
RTP Packet Loss (per RTCP packet)	548	0.000	0.000	0.000
RTP Jitter (msec, per RTCP packet)	548	0.000	0.002	1.000

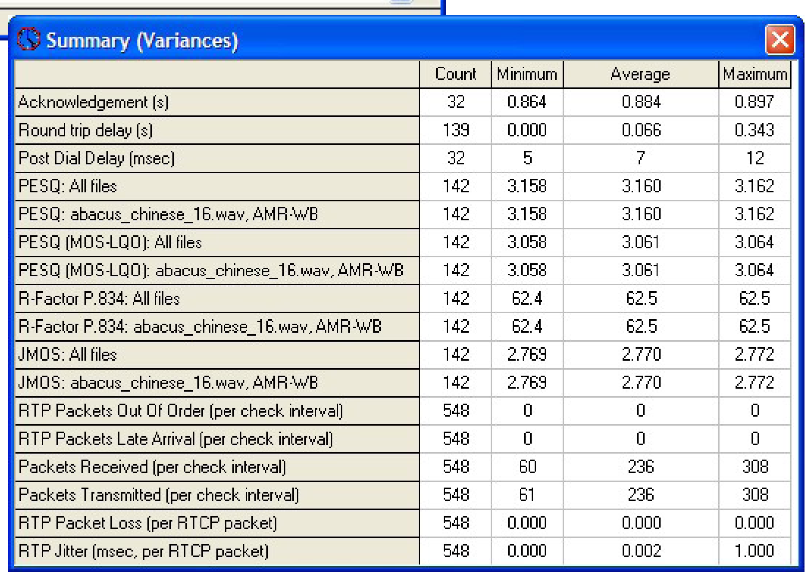
SPIRENT®

VOICE QUALITY

- G.107 The E-Model, a Computation Model for Use in Transmission Planning
- P.830 Telephone Transmission Quality: Methods for Objective and Subjective Assessment of Quality—Subjective Performance Assessment of Telephone-Band and Wideband Digital Codecs
- P.861 Objective Quality Measurement of Telephone-Band (300–3400 Hz) Speech Codecs
- P.862 Perceptual Evaluation of Speech Quality (PESQ): An Objective Method for End-to-End Speech Quality Assessment of Narrow-Band Telephone Networks and Speech Codecs
- P.862.1 Mapping Function for Transforming P.862 Raw Results Score to MOS_LQ
- Recommendation P.862.2—Wideband extension to Recommendation P.862 for the assessment of wideband telephone networks and speech codecs
- J-201.01v1 A Method for Speech Quality Assessments of IP Telephony, Japanese MOS



	Total	1	2	3	4	5	6	7
Script attempts	32	1	0	1	0	1	0	1
Script completions	0	0	0	0	0	0	0	0
% Script completions	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Call attempts	64	1	1	1	1	1	1	1
Call attempts per second (average)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Call attempts per second (momentary)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Call completions	64	1	1	1	1	1	1	1
% Call completions	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0
Registration attempts	0	0	0	0	0	0	0	0
Registration attempts per second (average)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Registration attempts per second (momentary)	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000	0.0000
Registration failures	0	0	0	0	0	0	0	0
Registration retry attempts	0	0	0	0	0	0	0	0



	Count	Minimum	Average	Maximum
Acknowledgement (s)	32	0.864	0.884	0.897
Round trip delay (s)	139	0.000	0.066	0.343
Post Dial Delay (msec)	32	5	7	12
PESQ: All files	142	3.158	3.160	3.162
PESQ: abacus_chinese_16.wav, AMR-WB	142	3.158	3.160	3.162
PESQ (MOS-LQO): All files	142	3.058	3.061	3.064
PESQ (MOS-LQO): abacus_chinese_16.wav, AMR-WB	142	3.058	3.061	3.064
R-Factor P.834: All files	142	62.4	62.5	62.5
R-Factor P.834: abacus_chinese_16.wav, AMR-WB	142	62.4	62.5	62.5
JMOS: All files	142	2.769	2.770	2.772
JMOS: abacus_chinese_16.wav, AMR-WB	142	2.769	2.770	2.772
RTP Packets Out Of Order (per check interval)	548	0	0	0
RTP Packets Late Arrival (per check interval)	548	0	0	0
Packets Received (per check interval)	548	60	236	308
Packets Transmitted (per check interval)	548	61	236	308
RTP Packet Loss (per RTCP packet)	548	0.000	0.000	0.000
RTP Jitter (msec, per RTCP packet)	548	0.000	0.002	1.000

SUPPORTED MODULES & PLATFORMS

Supported Platforms:

- Abacus 5000 IP Telephony Migration Test System

Supported Module:

- IP Telephony Signaling and Media Traffic Generator (ICG3D)

REQUIREMENTS

- Abacus 5000 13-slot (P/N SPT-3150), or 4-slot (P/N SPT-3050), or 3-slot (P/N SPT-3040) chassis and ICG3D (P/Ns ICG-3000D, ICG-3001D, ICG-3200D, and ICG-3201D)
- Windows® XP Professional SP2 or Windows Vista® Business operating systems
- 3.0 GHz Pentium® 4 or equivalent with 1 GB or RAM
- 2.5 GB of available disk space
- One Ethernet cable and one 10/100/1000 Mbps Ethernet card installed in the PC
- Sound card and speakers for Audio monitor (listen to any 2 channels from the controlling PC)

ORDERING INFORMATION

- SIP & IMS Package, ICG3: Includes SIP, G.723, G.726, IPv6, E-Model, SIP Scripting, DHCP, IMS Security, SigComp (P/N SWF-3290)
- All the ordering information for the Abacus 5000, and the ICG3 circuit generator with firmware options is available in the following data sheets:
  - Abacus 5000—IP Telephony Migration Test System
  - ICG3 Subsystem—IP Telephony Signaling and Media Traffic Generator

SPIRENT SERVICES

Abacus 5000 and Abacus 50 test systems come with comprehensive warranty, maintenance and support packages with Spirent Communications' full commitment to helping you get the most from our innovative technology.

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 Website at www.spirent.com/gs or contact your Spirent sales representative.

AMERICAS 1-800-SPIRENT • +1-818-676-2683 • sales@spirent.com

EUROPE AND THE MIDDLE EAST +44 (0) 1293 767979 • emeainfo@spirent.com

ASIA AND THE PACIFIC +86-10-8518-2539 • salesasia@spirent.com