SPIRENT Communications

Inspired Innovation

Abacus™ 5000 IP Telephony Signaling and Media Traffic Generator

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

The Abacus 5000 ICG3 subsystem simulates VoIP calling functionality by originating and terminating calls in a single card. The ICG3 subsystem performs high stress/performance signaling and RTP media testing.

APPLICATIONS

- Characterize VoIP gateways, VoIP PBXs, gatekeepers, proxy servers, media gateway controllers, softswitches and other internetworking gateways and PSTN network elements before trial
- Validate signaling system performance with high volume SIP signaling testing
- Identify RTP capacity limits, functionality, performance and interoperability
- Measure voice quality based on established standards for MOS, PSQM, PSQM+, PESQ-WB, MOS-LQO, R-Factor (P.834), J-MOS, and E-Model – R-Factor (G.107)
- Perform end-to-end service assurance testing for interfaces between the traditional PSTN and the VoIP packet based network
- Perform QoS of IVR and Voice Mail servers with Scripting for Voice Pattern Matching
- Simplify the testing of IMS and fixed to Mobile convergence

The ICG3 subsystem simulates multiple IP telephones and/or gateways, generating the call signaling and delivering the signaling (up to 20K channels for SIP) and/or media (up to 10K RTP calls for SIP with G.711) traffic to a system under test.

The ICG3 subsystem tests VoIP gateways, VoIP PBXs, gatekeepers, proxy servers, media gateway controllers, softswitches and other internetworking gateways and PSTN equipment.

The ICG3 subsystem executes endpoint registration requests – up to 65K simultaneous registered SIP users per ICG3 subsystem – to measure the capacity of SIP proxy servers.

Spirent has the most complete TDM, VoIP and analog solution in one platform using the same GUI. For TDM: OCG3, TCG3, PCG3, Abacus 50 T1/E1. For VoIP: ICG3, Abacus 50 Ethernet Test System. For analog: XCG3, ECG3, Abacus 50 Analog and Abacus 100 Analog.

ICG3 Front Card

IDI3 Rear Card

BENEFITS

- Simplify the testing of IMS converged IP telephony and PSTN networks and services with functional and performance testing for SIP, SIP-T, SIP-I, Skinny, H.323, MGCP/NCS, H.248/Megaco, SIGTRAN M3UA/M2UA/IUA, BICC, clear channel, softswitch emulation and PSTN/IP ladder diagram
- Achieve overall cost savings by giving the user full flexibility in convergence testing with synchronized IP, TDM and analog measurements with the same user interface

ICG3 SUBSYSTEM FEATURES

Voice Quality Testing

- Perform voice quality measurements on 256 channels using MOS, PSQM, PSQM+
- Perform voice quality measurements on 128 channels using PESQ, MOS-LQO, R-Factor (P.834), J-MOS
- Perform voice quality measurements on 64 channels using PESQ-WB
- Perform voice quality measurements on thousands of RTP streams using E-Model – R-Factor (G.107)

Quality of Service (QoS) Testing

- Perform QoS validation on 128 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

- VoIP call generation for SIP, Skinny Client Control Protocol (SCCP), H.323, MGCP/ NCS, H.248/Megaco, SIP-T, SIGTRAN (M3UA/ M2UA, IUA-PRI/IUA-BRI), BICC, T.38 fax over IP, T.30 Fax, 3PCC, H.235-GRQ, clear channel
- Generate or terminate up 20K channels of signaling without voice per subsystem for SIP only (ICG3D)
- Generate or terminate up to 4,096 channels of signaling without voice per subsystem for other VoIP protocols
- Generate or terminate up to 8K media calls (RTP and signaling for SIP only) with path confirmation at packet size of 20 mS on ICG3D
- Generate or terminate up to 4,096 media calls (RTP and signaling for other VoIP protocols except SIGTRAN) with path confirmation
- Generate call setups to media gateway controllers, gateways, or softswitches
- Load profiling (Saw Tooth, Rectangle, Trapezoid and Poisson)
- Network Topology Diagram with predefined diagrams

Troubleshooting and Diagnostics Capabilities

- Verify that 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, calls 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 the 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
- Multi/Dual user
- Graphical display of Measurements-over-Time
- Measure one-way delay measurements
- Softswitch emulation for testing new generation of signaling gateways and trunking gateways
- PSTN/IP ladder diagram displays synchronized information available in a multi-ladder diagram for SS7, H.248/ Megaco and MGCP
- Up to 13 ICG3 subsystems per chassis
- 2-port with dual media Gigabit Ethernet for originating and terminating calls on 1 ICG3/ICL for all VoIP protocols

- 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), EVRC-B, 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 for SIP 65K simultaneous registered SIP users
- Generate or terminate up 20K channels of signaling without voice
- Generate or terminate up to 8K media calls (RTP and signaling for SIP) with path confirmation at packet size of 20 mS on ICG3D
- 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
- 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 1,000 SIP endpoints
- Gradual increase RAPS over time
- DHCP for IPv4
- SIP Proxy emulation, registrar and call routing
- RTP Replay
- IMS Security (IPSec and AKA)
- SIP Signaling Compression (SigComp)
- SIP Supplementary Feature Testing
- SIP Trunking
- Mobile SIP UE
- SIP Loopback

SUPPORT FOR SIP-T AND SIP-I

- Advanced trunking signaling testing 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

- Up to 65K H.323 end-point registrations per subsystem
- Send bearer capacity information within “speech” using H.323
- Support H.323 IRR call status report
- T.38 fax over H.323
- Configurable GRQ (Gatekeeper ReQuest) - 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
- Call Tracer (Ladder diagram)
- Multiple VLAN/MAC configuration
- ETSI H.248/Megaco
- H.248/Megaco over SCTP
- 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
- Test VoIP Security with Secure RTP
- Call Tracer (Ladder diagram)
- Multiple VLAN/MAC configuration
- Message Fragmentation and 521 redirect
- Hook-Flash

SIGTRAN SUPPORT

- Advanced trunking signaling testing with SIGTRAN M3UA and IUA
- Softswitch emulation for SIGTRAN M3UA and IUA
- Support for SCTP failover

BICC SUPPORT

- BICC over SCTP/IP
- RTP media for BICC over SCTP

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 measurements
- E-Model – R-Factor (G.107)

Call Routing/Switching

- Routes and switches H.248/Megaco, and MGCP calls
- Detects of hook-off on the originating side
- Collection of digits dialed by the originating subscriber
- Translation of 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

- 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
- 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

Echo Measurements

- Echo cancellation on/off
- Echo delay
- ERL (Echo Return Loss)
- ERLE measurement (Echo Return Loss Enhancement)
- TELR measurements (Talk Echo Loudness Rating)
- Support echo measurements on 8 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 Recommendation H.248 (now H.248.1) in the ITU-T
- SIP-T RFC 3372
- SIGTRAN M3UA RFC 2905, RFC 3332
- SIGTRAN IUA, RFC 2960, RFC 3057
- 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
- ITU-T Q.1902.4 BICC (CS2) Basic Call Procedure

Interfaces

- ICL/ICG3 subsystems: 1- or 2-port dual media Gigabit Ethernet with standard 10/100/1000 Base-T and optional SX/LX transceivers

PHYSICAL CONNECTION

- ICG3 and ICL front card fits into one Abacus 5000 slot
- IDI3 dual media Gigabit Ethernet rear card provides standard two RJ-45 connectors with optional SFP connectors
- The optional 1000Base-SX Gigabit Ethernet SFP transceiver supports the following fiber cables: 2 to 550 meters with 50/125 μm fiber and 2 to 275 meters with 62.5/125 μm fiber
- The optional 1000BASE-LX Gigabit Ethernet SFP transceiver supports single mode 1310nm

ELECTRICAL

- Power draw: Approx. 50 W per board

ORDERING INFORMATION

IP Telephony Subsystems

- ICL3 1-port dual-core – VoIP call RTP performance - no DSP (P/N ICG-3000D)
- ICG3 1-port dual-core – VoIP call RTP performance – with DSP (P/N ICG-3001D)
- ICL3 2-port dual-core – VoIP call RTP performance – no DSP (P/N ICG-3200D)
- ICG3 2-port dual-core – VoIP call RTP performance – with DSP (P/N ICG-3201D)

Rear Card

- 2-Port 10/100/1000 copper rear card (add SFP for dual media) (P/N ICG-IDI3)

Transceivers

- 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)

Firmware Options

- SIP for VoIP call control on ICL/ICG3 (P/N SWF-3001)
- MGCP/NCS for VoIP call control on ICL/ICG3 (P/N SWF-3002)
- H.248/Megaco on UDP for VoIP call control on ICL/ICG3 (P/N SWF-3003)
- H.323 for VoIP call control on ICL/ICG3 (P/N SWF-3004)
- SIP-T, SIP-I call control, ICL/ICG3 (P/N SWF-3005)
- PSQM, PSQM+ on ICG3 (P/N SWF-3009)
- PESQ on ICG3 (P/N SWF-3010)
- SIGTRAN (M3UA,SCTP), ICL/ICG3 (P/N SWF-3013)
- SIGTRAN (M2UA/SCTP), ICL/ICG3 (P/N SWF-3138)
- SIGTRAN (IUA,PRI), ICL./ICG3 (P/N SWF-3126)
- SIGTRAN (IUA,BRI), ICL/ICG3 (P/N SWF-3137)
- Video Encoding (H.263) on ICL/ICG3 (P/N SWF-3014)
- T.38 Fax over ICL/ICG3 (P/N SWF-3111)
- T.30 Fax up to V.17, ICG3 (P/N SWF-3122)
- IPv6 (for SIP and RTP), ICG3 (P/N SWF-3125)
- Call routing/switching, ICL/ICG3 (P/N SWF-3128)
- RTP Video (H.264), ICL/ICG3 (P/N SWF-3129)
- Support for 2-port ICG3/ICL (P/N SWF-3130)
- Stacked VLAN, QinQ, ICL/ICG3 (P/N SWF-3131)
- SRTP, ICL/ICG3 (P/N SWF-3132)
- TLS, ICL/ICG3 (P/N SWF-3133)
- E-Model, ICL/ICG3 (P/N SWF-3134)
- SIP Scripting, ICL/ICG3 (P/N SWF-3135)
- 2x Turbo RTP, ICL/ICG3 (P/N SWF-3136)
- Clear channel for ICL/ICG3 (P/N SWF-3203)
- SCCP-Skinny Client Control Protocol, ICL/ICG3 (P/N SWF-3215)
- Echo measurements, ICG3 (P/N SWF-3226)
- Scripting for Voice Pattern Matching, ICG3 (P/N SWF-3239)
- H.248/Megaco over SCTP, ICL/ICG3 (P/N SWF-3241)
- BICC (P/N SWF-3242)
- RTCP on Turbo RTP, ICL/ICG3 (P/N SWF-3243)
- SIP Registrar (P/N SWF-3244)
- DHCP, ICL/ICG3 (P/N SWF-3248)
- PESQ-WB (P/N SWF-3249)
- SIP IMS Security - IPSec and AKA (P/N SWF-3252)
- AMR-WB (G.722.x)Encoding and Decoding, ICG3 (P/N SWF-3253)
- EVCR-B Encoding and Decoding, ICG3 (P/N SWF-3254)
- SIP Proxy Emulation, ICL/ICG3 (P/N SWF-3255)
- SIP Proxy Emulation Compression (SigComp), ICL/ICG3 (P/N SWF-3256)
- iLBC Encoding and Decoding, ICG3 (P/N SWF-3259)
- GSM EFR, FR Encoding and Decoding, ICG3 (P/N SWF-3362)
- RTP Replay, ICL/ICG3 (P/N SWF-3363)
- G.723, G.726 Encoding and Decoding, ICG3 (P/N SWF-3364)
- Mobile NB Encoders and Decoders GSM-AMR, EVRC, ICG3, (P/N SWF-3365)
- SIP and IMS Package, ICG3 (P/N SWF-3290)

FOR MORE INFORMATION

Visit Spirent Communications' Website 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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