SPIRENT TESTCENTER

SIP FOR TELEPRESENCE AND VOLTE

Delivering next-generation unified communication services includes packaging voice and video into rich communication suites (RCS) of application. Service Providers use Session Initiation Protocol (SIP) to converge these applications into simple, cost-effective RCS. These applications are typically built into SIP UEs of all kinds, from desktop PC applications to IP-PBX phones to CPE devices to 3G/4G mobile UEs including VoLTE.

APPLICATIONS

The Spirent TestCenter™ SIP solution, based on RFC 3261 and 3GPP IMS specifications, enable 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, VoIP firewall, and video teleconference and telepresence (TIP) exchange servers. Spirent TestCenter SIP includes integration of various IP protocols, such as DNS and DHCP, to realistically emulate a typical SIP UE, including generic SIP UE, Cisco Telepresence UE and/or VoLTE UE.

Whether in a controlled lab environment, a pilot, or a live operational network—Spirent TestCenter SIP applications measure and validate key control-plane signaling delays to ensure high user quality of service by emulating thousands of multimedia SIP UAs simulating voice and/or Telepresence calls during busy hours.

Spirent TestCenter SIP, along with RTP/RTCP, can be used to validate the KPI (Key Performance Index) of voice and video media elements, such as IP media gateways, IP PBX, Media Servers (IVR and Voicemail), Cisco Telepresence Exchange Servers/Call Managers, and IMS P-CSCF servers and MRF.

All real-time KPI and delay measurement results enable service providers and NEMs to quickly address multiple media service issues and reduce network downtime. Combined with broad Layer 2-3 test capabilities, Spirent TestCenter SIP is the test solution for pre- and post- VoIP and teleconferencing deployment testing for NEMs, large carriers, wireless providers, MSOs, enterprise PBX, and cloud-based VoIP service providers.

❖ SIP/IMS application server testing for verifying SIP Proxy/Registrar server functions, SIP registration and session setup and teardown, and registration subscription
❖ Testing Cisco Telepresence (TIP) Call Managers and Encoding gateways with H.264 video and high fidelity audio codecs
❖ High-scale VoIP user (SIP UA) emulation and call generation, including VoLTE UE emulation
❖ Evaluating key performance parameters of triple-play service infrastructure by emulating thousands of VoIP users (SIP UAs) along with other services and protocols on a single port
❖ A rich set of real-time results on both SIP signaling and Simulated RTP/RTCP data streams are available
❖ Measuring subjective MOS quality scoring with VQMon (ITU P.564) using media-encoded RTP streams generating real-world media traffic
❖ Spirent TestCenter helps users set up thousands of SIP UAs and VoIP calls with easy-to-use wizards
FEATURES AND BENEFITS

- **Mobile VoIP UE Emulation**: Emulates 3GPP SIP Mobile UE for UMTS and GPRS profiles.
- **VoLTE UE Emulation**: Supports VoLTE Mobile UE with AMR-NB (narrowband) and AMR-WB (Wideband) codecs. Assess voice quality of VoLTE services over topology emulation, simulating calls over different networks.
- **Simulates Telepresence Endpoints**: Supports SIP-based signaling for registration, establishment and teardown of Telepresence calls. Emulates Cisco CTS500 endpoints with 720p video resolution.
- **Media Codecs and Speech Payload**: Supports standard G.711 A/µ-law, G.729, G.726, G.723.1, and G.728 speech-encoded audio samples. Media payload in RTP streams can be customized with random data allowing more flexibility controlling media size.
- **Voice Quality Analysis (VQA)**: Connection rate, number of sessions, and SIP delay metrics do not tell the whole story about voice quality. The VQA module, based on P.564, provides voice quality metrics, such as MOS, R-Factor, jitter, packet loss, packet delay, and other voice and network-transport metrics to further provide accurate analysis of the ability of the IP network to provide high QoS VoIP services.
- **Simulated and Encoded RTP**: Simulated RTP allows specific use case where accurate delay measurements are required for verifying minimal delay in transporting real time VoIP traffic. Encoded RTP is used for verifying voice quality of encoded media/speech over IP.
- **End-to-End or End-to-DUT calls**: Calls can be established STC end-to-end for network testing or STC end-to-real SIP UA (DUT) for subjective test analysis.

TECHNICAL SPECIFICATIONS

- **SIP UA (host) configuration**
  - General host configuration
  - Local SIP Port
  - Ring duration in sec
  - SIP compact format
  - UA number format with step
- **Proxy/registration servers configuration**
  - Proxy/Registration servers IPv4 and IPv6
  - Proxy/Registration servers port addresses
  - Proxy/Registration servers domain names
- **Signaling Configuration**
  - SIP long and compact headers
  - Call ID domain name
  - IMS Secure Request URI
  - Anonymous call
  - IPv4 TOS/DiffServ and IPv6 traffic class
- **Audio/Video Configuration**
  - Audio Only
  - Audio and video
  - Telepresence
  - Signaling only (SIP sessions only, no RTP)
  - Audio and video UDP port addresses
  - Audio dynamic payload type
  - Video dynamic payload type (Telepresence only)
- **SIP RFC Extensions**
  - RFC 3261 and 3GPP TS 24.229
  - RFC 2327 SDP: Session Description Protocol
  - RFC 2617 HTTP Authentication: Basic and Digest Access Authentication
  - RFC 3087 Control of Service Context using SIP Request-URI
  - 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 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 4028 Session Timers in the Session Initiation Protocol (SIP)
  - RFC 3966 The tel URI for Telephone Numbers
SPIRENT TESTCENTER

SIP FOR TELEPRESENCE AND VOLTE

REQUIREMENTS

An IBM® compatible PC must meet the following minimum requirements to run the Spirent Avalanche:

- One 10/100/1000Base-T unshielded twisted pair (UTP) cable
- One 10 Mbps or 10/100/1000 Mbps Ethernet NIC card
- Intel® E6300 Core 2 Duo 4 (or equivalent)
- One serial port
- Minimum of 2GB of RAM
- Minimum 10 GB free space on the hard drive
- Windows® XP operating system, Service Pack 2 (SP2)
- An SVGA color monitor (or equivalent) and a mouse
- Microsoft Terminal Services™ or HyperTerminal™, or equivalent communications software package
- CD-ROM drive

ORDERING INFORMATION

<table>
<thead>
<tr>
<th>Description</th>
<th>Part Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP/RTP/RTCP BASE PACKAGE A</td>
<td>BPK-1060A</td>
</tr>
<tr>
<td>SIP ENCODED RTP &amp; SPEECH VQA BASE PKG A</td>
<td>BPK-1303A</td>
</tr>
<tr>
<td>SIP ENCODED RTP &amp; SPEECH VQA BASE PKG A - 2PK MOD</td>
<td>BPK-1303A-2XMOD</td>
</tr>
<tr>
<td>UPGRADE SIP ENCODED RTP &amp; SPEECH VQA PKG A 2PK TO FULL CHS</td>
<td>UPG-BPK-1303A-2XMOD</td>
</tr>
</tbody>
</table>

SPIRENT 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 website at [www.spirent.com](http://www.spirent.com) or contact your Spirent sales representative.