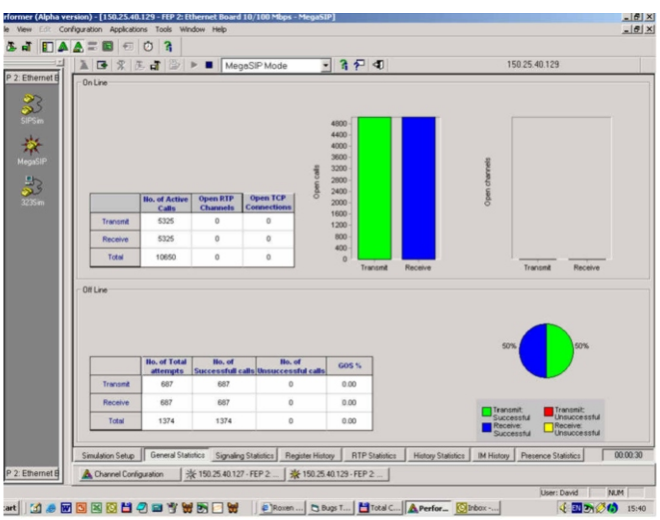
SIPSim

SIP-Based Services Simulation

SIP is gaining momentum by delivering market-driven applications as part of the IMS (IP Multimedia Subsystem) for 3G Cellular networks and beyond, validated by the 3GPP, 3GPP2 and OMA standardization. In addition, SIP is becoming dominant as the "Triple play" offers cutting edge services for broadband networks. As a result there is a growing need for advanced service testing in wireline and next generation cellular networks.

SIPSim™, a Services over IP (SoIP) Performer™ component, is an advanced hardware and software solution that generates high volume SIP-based services for SoIP systems testing, combining both signaling and media. SIPSim emulates real-world network conditions and enables developers, QA labs and Service Providers to benchmark, load test and verify proper protocol implementation in SoIP equipment, as well as ascertain compliance to standards, signaling integrity and Grade of Service (GoS). By emulating a virtually unlimited number of SIP terminals that can initiate one or more SIP sessions, receive network SIP responses and terminate existing calls, the SIPSim is capable of stressing different network elements such as:

- SIP entities: Proxy, Registration, Redirect servers, and IMS components (such as P-CSCF)
- End entities: Media Gateways, Trunking Gateways and IP PBXs/Phones
- Services entities: Presence and Instant Messaging (SIMPLE), PoC and Video servers
- Security and Administration entities: Session Border Controllers, (AAA) and firewalls/NAT servers.
- Broadband networks based on Cable, xDSL and WiMax networks.



Traffic Statistics

Portable Performer

Highlights

- RFC 3261 compliant.
- Initiates, responds to and terminates thousands of SIP calls and UACS.
- Register load testing (REGISTER and re-REGISTER).
- Provides fine-grain control over call setup rate, call length and other parameters.
- Transmits RTP voice packets encoded in various codec formats for load testing:
  G711 Alaw and Ulaw, G.722, G.723 (5.3 Kbps), G.723 6.4 Kbps), G.728, G.729A, AMR, H.263, MPEG4 and H.264 in addition to the DTMF according to RFC 2833.
- Enables the user to define groups of SIP entities using different call flows by:
  - Selecting a flow from a library of SIP flows for incoming and outgoing calls.
  - Updating the flow properties (such as provisional response).
  - Customizing SIP request messages and responses including the body and SDP.
- Generates a peak load of up to 2 million signaling calls simultaneously and up to 200,000 simultaneous calls with RTP for each Performer system using an open rate of 8,000 cps and 12 million calls per busy hour.
- Enables the user to define and integrate their own SIP flow to SIPSim's flow library.
- Enables emulating a wide environment using up to 40,000 multiple IPs (signaling and media) and up to 400,000 URIs per Performer system.
- Supports emulation of up to 4095 private networks using VLAN Tags.
- Displays call activity and completion statistics such as setups, releases, online active calls and GoS.
- Performs digest authentication.
- Supports the ENUM address method.
- Simulates predictive test conditions using 3 different traffic profiles: RAMP, Poisson and Normal distribution.
- Provides a fault insertion testing capability.
- Supports objective voice quality measurements together with other Performer components.
- Provides automatic test capabilities through MasterScript, a powerful scripting tool.

RADCOM

TEST-OF-THE-ART

SIPSim

Signaling Stress Capabilities

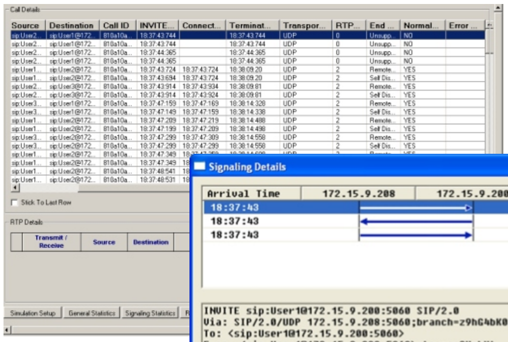
SIPSim's Signaling Stress is capable of generating peak loads with 500,000 simultaneous calls per segment, three million sessions per busy hour and a setup rate of 2,000 calls per second, fully scalable to multiple links. This capability can be used to test whether equipment such as SIP proxies, register servers, mediators or gateways can sustain high volumes of calls, in addition to determining their level of reliability and the time required to accept new calls.

SIPSim's Signaling Stress capability brings the SIP simulation package to the level of a high-end product. Its main benefits include high volume call generation, integration into the Performer and the control offered by the MasterScript tool.

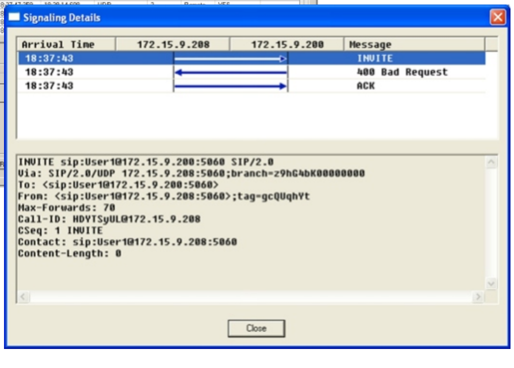
SIPSim can be used when stress testing is required for checking SIP proxy stability and validating the number of calls the proxies process.

Signaling Stress Highlights

- Offers scalable multi-server solution enabling an unlimited number of signaling calls/BHCA
- Generates peak loads with 500,000 calls simultaneously per segment
- Provides a setup rate of 2,000 calls per second per segment
- Generates over 3,000,000 BHCA calls per segment where each Performer system includes up to 4 segments
- Simulates predictive test conditions with various traffic models: Ramps, Poisson and Normal Distribution
- Displays online and history statistics
- Runs with other Performer applications such as MediaPro, QPro and Capture
- Emulates up to 100,000 URIs per segment, using up to 10,000 multiple IPs
- Provides full analysis of abnormal calls including the signaling flow



History Results

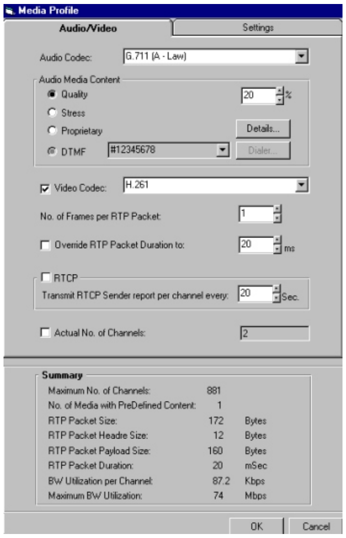


Abnormal Signaling View

Media Capabilities

The SIPSim capability is designed to generate peak loads with up to 50,000 RTP sessions per segment, in various codecs, for SIP calls created by SIP signaling, fully scalable to multiple segments. Utilizing RADCOM's hardware transmission capabilities, the SIPSim capability enables testing and verification of media gateway, mediator or media server behavior and reliability with high volumes of calls. This feature is particularly useful when testing network readiness for SIP solutions or large-scale gateways between packet- and circuit-switched networks and media servers or administration devices such as Session Border Controller and firewalls.

To be successful in the VoIP industry, equipment vendors must deliver high quality voice without significantly impacting data delivery in converged networks. For example, a gateway vendor must be able to guarantee solid performance even under the most adverse network conditions. Such a vendor wants to see the quality of media after it goes through the gateway. In these cases, they can use the MediaStress capability to generate thousands of calls with RTP through the gateway.



Media Profile Configuration

Media Capabilities Highlights

- Generates up to 4,500 outgoing RTP channels per Fast Ethernet link and up to 50,000 outgoing RTP channels per Giga Ethernet link
- Generates RTP channels at a very high level of accuracy, using RADCOM's proprietary hardware
- Transmits predefined content for voice quality evaluation (PAMS/PESQ) or video, including DTMF (RFC 2833)
- Enables testing of QoS mechanisms by configuration of different Type of Service (ToS) definitions for RTP streams
- Enables control of RTP packet structures such as number of frames per packet and RTP packet duration
- Allows configuration of RTCP packet transmit intervals, providing information about the delay, jitter and packet loss
- Displays call statistics, distinguishing between successful and failed calls

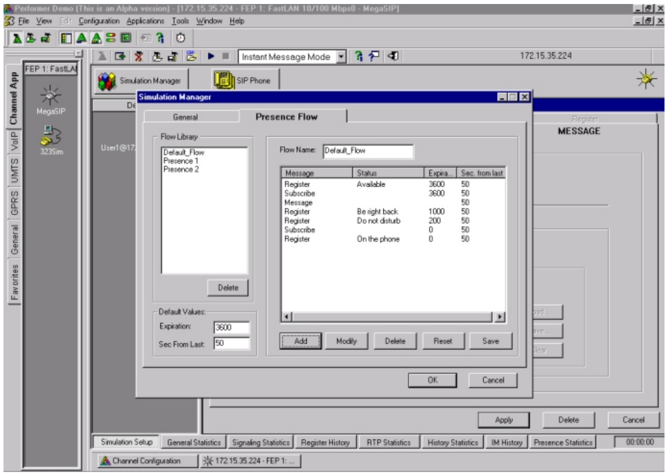
Presence and Instant Messaging

The Presence and Instant Messages (IM) market is growing rapidly. IM is the first presence functionality application, which expands to many other services. A Presence and Instant messaging system allows users to subscribe to each other and be notified of changes in state, as well as send short instant messages.

The user's ability and willingness to be reached for communication is defined by an information set known as presence information. Publishing the presence information is done by sending a SIP REGISTER message. Users may then subscribe to a service through the SUBSCRIBE message and can ask to be notified immediately of changes in the presence information of the service for a duration of time. With the SIPSim, two types of testing are performed:

Stress testing of the Presence server by creating different flows of messages such as REGISTER, UNREGISTER, SUBSCRIBE, UNSUBSCRIBE and MESSAGE.

Stress testing of the IM server by sending a REGISTER, SUBSCRIBE and then many MESSAGE methods every fixed time interval.



Presence and Instant Messaging Configuration

Push-To-Talk/PoC

Instant Talk over Cellular (PoC) is a popular two-way radio ("walkie-talkie") service available through cellular phones, enhancing cellular services and bringing new business opportunities into the domain of real-time voice communications. PoC calls are a one-way communication: while one person speaks, the other listens. The turns to speak are granted by pressing the Push-to-Talk key on a first come, first served basis within the talk group.

SIPSim supports PoC call simulation by configuring its end-point terminals to behave as a PoC entity. It generates the calls through the Early Media call flow scenario, which starts sending media at the earliest time possible, thus saving the user a long set-up call.

Functionality Testing Capabilities

SIPSim provides deep functionality testing capabilities that enable the user to test the functionality of DUT / NUT using different scenarios and stress level conditions.

As part of the wide range of network emulation, SIPSim enables users to create groups of multiple users and for each group to define a profile of behavior that includes:

- User registration call establishment procedure
- Answering policy (use of provisional response, answer all, reject, etc.)
- Customization of all SIP requests and responses
- Definition of incoming flow and outgoing flow from a library of call flows based on:
  - Predefined flows, including supplementary services (hold, transfer…), call redirect and erroneous flows
  - User-defined flows, where the user can integrate flows to SIPSim using a special SDK environment

SIP Services

SIP is an emerging IP-based protocol that is critical for deploying converged and next-generation real-time voice, data and video communication services. Its flexible and open service creation environment, offers users more control to easily create and quickly deliver profitable, next generation services to multiple end-points across wireless and wireline networks.

SIPSim enables users to test services such as Instant Messaging and Presence, IVR, instant Talk over Cellular, Voice and Video media (PoC).

Voice and Video

SIPSim enables developers and service providers to benchmark, load test and verify proper protocol implementation in SIP-compliant, Voice and Video over IP equipment. Working in conjunction with other Performer components, it offers users add-on measurements such as QoS, GoS and voice quality evaluation.

The SIPSim transmits RTP voice packets encoded in various codec formats for load testing: G711 Alaw and Ulaw, G.722, G.723 (5.3 Kbps), G.723 (6.4 Kbps), G.728, G.729A and AMR. Video codecs include H.263, MPEG-4 and H.264. DTMF according to RFC2833 is also supported.

SIPSim

The Performer

The Performer is a comprehensive solution for pre- and post-deployment stages, R&D verification, stress testing, troubleshooting and recurring IPMM, Cellular and VoD system performance testing. It generates realistic network environment stress levels on new VoD devices and applications, and then tests the quality and grade of service delivered. Additionally, the Performer provides session-oriented Consultants, using a wide range of interfaces based on GEAR. GEAR is a proprietary generic analyzer processor chip offering hardware-based full-line rate analysis capabilities at up to 2.5 Gbps.

Services over IP Performer Suite

The SoIP (Services over IP) Performer is a comprehensive IP test solution integrating GEAR-based hardware and software. Controlled from an easy-to-use console, the complete Performer SoIP suite includes the SIPSim™, 323Sim™, SoIP call generator, MediaPro™ session-oriented SoIP consultant and QPro™ voice quality evaluation tool.

Cellular Performer Suite

Cellular Performer is a comprehensive cellular test solution for vendors, QA and integration labs, R&D and operators. Based on the field-proven Performer platform, it integrates RADCOM's proprietary GEAR. The complete Cellular Performer suite offers a range of applications for troubleshooting 2.5 and 3G networks, including GPRS, UMTS and CDMA2000.

Performer Analyzer

The Performer Analyzer is a comprehensive datacom test solution. Based on the field-proven Performer platform, it integrates RADCOM's proprietary GEAR which provides hardware-based full-line rate capture and analysis at up to 2.5 Gbps.

The Performer Analyzer uses RADCOM's proprietary powerful hardware-based GenFEP, offering a flexible and upgradeable solution as a well as a wide range of full-line rate technology and protocol-independent capabilities. The complete Performer Analyzer suite offers a wide range of technologies and applications for analyzing, decoding and troubleshooting datacom networks.

Target Users

Developers use SIPSim to analyze equipment under stress, verify standard compliance and assess signaling integrity during development stages.

QA labs use SIPSim to stress equipment in order to substantiate Quality of Service and voice quality.

Service providers and carriers use SIPSim to evaluate different systems, analyze price-performance, verify system integrity for the deployment stage and confirm standard compliance verification for interoperability issues.

3G cellular operators use SIPSim to evaluate and employ IP services in their cellular networks.

Field service personnel use SIPSim to identify and troubleshoot network segments that may affect audio quality.

Specifications

Performer Servers

R1000, Rack-mount 2U, Performer Server (single segment)
Number of FEPs: Up to 2 plus Sync cards
Dimensions: w x d x h: 440 x 470 x 89 mm (17.5 x 18.7 x 3.5 in)

R4000, Rack-mount 5U, Performer Server up to 4 segments
Number of FEPs: Up to 8 plus Sync cards
Dimensions: w x d x h = 430 x 680 x 220 mm (17 x 27 x 8.7 in)

P1000, Portable Server
Number of FEPs: Up to 3, plus Sync cards
Dimensions: w x d x h: 360 x 480 x 130 mm (14.17 x 18.9 x 5.11 in)

Performer Console

PC: Pentium IV 1.4 GHz, 512 MB RAM or more (recommended)
Monitor: VGA 1024 x 768
Hard disk: Minimum 4 GB free for program files
At least 2 GB recommended for data storage
Operating system: Windows 2000/XP

Ordering Information

PA-SW-SIPSim
PA-SIPSIG-High
PA-SIPSIG-Medium
PA-SIPMedia-Basic
PA-SIPMedia-Medium
PA-SIPMedia-High
PA-SIP-Video
PA-SIP-Extra-users

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