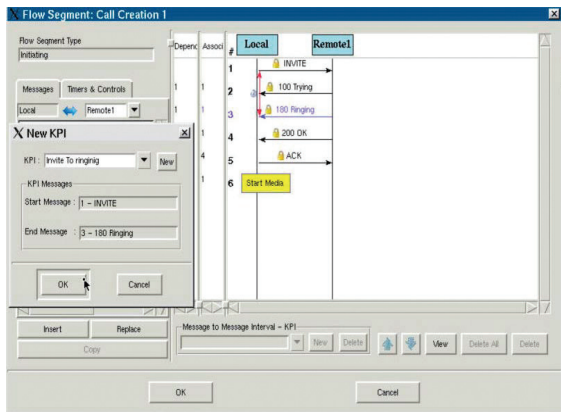
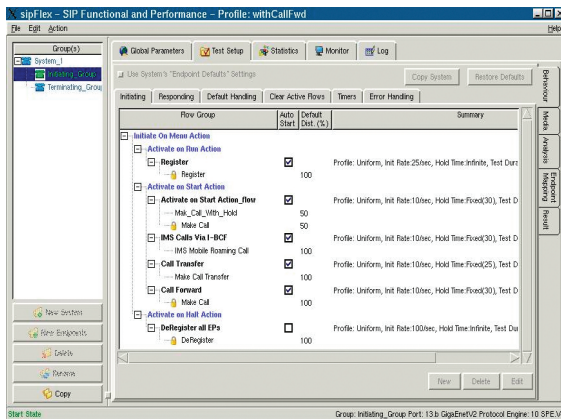


IP-to-IP Gateway Test Suite



Highlights

- 128 000 RTP streams
- 256 000 endpoints
- 2000 sessions per second with RTP
- Theft of service and session policing tests
- Media-pinhole opening and closing tests
- Generation and analysis of voice, video and DTMF services
- QoS and VQT testing
- Video quality assessment with MDI
- Extensive list of codecs
- Comprehensive statistics
- Transport layer security (TLS)
- IPSec
- IKEv1
- SIP and H.248

Overview

IP-to-IP gateways such as session border controllers, H.248 border gateways, security gateways and firewalls, play a critical role in enabling and protecting the new generation of multimedia services, which are driving the deployment of VoIP and fixed mobile converged (IMS) networks.

VoIP and IMS IP-to-IP gateways support a number of core functions such as security against theft of service and malicious attacks using SIP flood/DoS and rogue media, per-session policing, QoS monitoring and reporting as well as secure access to services. These functions are critical to ensure the viability of VoIP and IMS services. This IP-to-IP Gateway Test Suite focuses on testing these functions as well as performance, scalability and reliability of the IP-to-IP gateways.

Large-Scale and High Performance Signaling Solution

This solution can simulate over 2000 sessions per second with signaling and RTP. It offers a large-scale test solution, which can emulate up to 256 000 endpoints (distributed over eight groups of endpoints) in subscriber and/or gateway modes with each subscriber supporting a unique IP address.

Comprehensive Media Generation and Analysis

The IP-to-IP Gateway Test Suite can generate and terminate up to 128 000 RTP streams on a single chassis, providing the ability to emulate up to 64 000 active sessions while generating RTP for the entire duration of each session.

Each session is unique and can support its own media characteristics such as codec type and packetization interval and wave file. RTP and path verification analysis can be performed on every established session to verify the integrity of the RTP packets that are received.

RTP (voice, video and DTMF) analysis (VQT, QoS, MDI) can be performed on up to 16 000 simultaneous sessions in real time for the duration of the sessions. The results of the analysis are displayed in real time and updated on a one-second interval.

Tightly-Coupled Signaling and Media

In real-world deployments, signalling and the associated RTP stream for each session is tightly coupled. The media will start only after signaling has been successfully established and will stop as soon as the session has been terminated from a signaling perspective.

The IP-to-IP Gateway Test Suite emulates real world implementations by guaranteeing a tight coupling between the signaling and media for each session, regardless of the session characteristics.

The time between signaling establishment and media generation and between signaling termination and media termination is of the order of milliseconds, guaranteeing that the device under test will not terminate the sessions because of inactivity after the sessions are successfully established.

The media start and stop times relative to session establishment and tear down can be configured.

Real-World Traffic Characteristics

In order to emulate real world traffic, the IP-to-IP Gateway Test Suite allows users to easily define the following traffic characteristics for up to eight independent groups of endpoints:

- Session establishment rate and distribution
- Session duration
- Media mix in terms of percentage of certain media types (audio, video and DTMF) and codecs where applicable
- Percentage of sessions with media – rest of sessions will be placed on hold
- Call flow and signaling messages content

QoS, VQT and DTMF Analysis

The IP-to-IP Gateway Test Suite provides the ability to measure the quality of the end-user experience by performing voice, video and DTMF analysis in real time on up to 16 000 simultaneous RTP streams on a single chassis.

Thresholds can be configured for a number of parameters. If the threshold conditions are not met, the session is declared failed and detailed information is provided at the endpoint group, media type (voice, video and DTMF), codecs and session levels.

The following media quality measurements are provided and updated in real time:

- QoS: packet loss, delay and jitter
- Voice quality: mean opinion score (MOS) and R-factor
- Video quality: media delivery index (MDI)

The application also supports generation and analysis of DTMF in-band and out-of-band using signaling (SIP info and H.248 notify/modify messages) and RFC 2833.

Session Policing and Protection Against Service Theft

A core function of IP-to-IP Gateway Test Suite is to perform per-session inspection to ensure that each session complies to the negotiated bandwidth. This function is critical not only to protect against theft of services (session negotiating audio but generating video for example), but also to ensure that sessions that comply to their negotiated contract are not impacted by non-conforming sessions.

A single non-compliant video stream of 10 Mbit/s could consume the required bandwidth of 156 G.711 64 kbit/s audio calls and if these calls are starved of their required bandwidth, the quality of the user experience will be significantly degraded.

The IP-to-IP Gateway Test Suite provides the ability to easily test this core function of IP-to-IP gateways. Users can simulate theft of services by selecting different media/codec types for negotiation and transmission.

The application will detect theft-of-service activities and provide detailed statistics in real time so that the user can verify whether the gateway detected and penalized the non-conforming media streams while the streams that conformed to their negotiated contract were not impacted.

Media Integrity and Routing Verification

Under load an IP-to-IP gateway could route packets to unintended subscribers, which could result in crosstalk.

The path and media verification tests of the IP-to-IP Gateway Test Suite can detect misrouted packets, mislabeled ToS and DSCP headers, unexpected media types and inactivity after call establishment.

Pinhole Opening and Closing Verification

IP-to-IP gateways provide access to the network by opening a media path typically referred to as a pinhole for each established session. Resources (memory, processing cycles, etc.) are allocated to each pinhole and the associated media stream. On termination of each session, the IP-to-IP Gateway Test Suite should deallocate the resources and close the pinhole.

Failure to close the pinhole and deallocate the resources will result in a resource-leak, which will eventually cause the gateway to run out of resources and crash. In addition, open pinholes that are not associated with an active session are security loopholes, which can be exploited by hackers to steal services and attack the network with rogue media.

The IP-to-IP Gateway Test Suite supports canned test scenarios to verify that pinholes are successfully opened when sessions are established and closed once the sessions are terminated. The solution also supports the ability to generate media through a pinhole after the session has been terminated and detect whether the media was forwarded by the gateway.

TLS Security

Each emulated SIP endpoint in the SIP IP-to-IP Gateway Test Suite can be TLS enabled providing the ability to emulate up to 256 000 unique security associations. The endpoints can emulate both client and server modes and support the following cipher suites:

- TLS_RSA_WITH_AES_128_CBC_SHA
- TLS_RSA_WITH_AES_256_CBC_SHA
- TLS_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_DHE_RSA_WITH_AES_128_CBC_SHA
- TLS_DHE_RSA_WITH_AES_256_CBC_SHA
- TLS_DHE_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_DHE_RSA_WITH_DES_CBC_SHA
- TLS_RSA_WITH_RC4_128_SHA

Bandwidth Thresholds Configuration

In a deployed network, the bandwidth utilization is a complex calculation which depends on the session establishment rates and hold-time and mix of media/codecs. This computation can be very complex especially in scenarios where the traffic patterns are non-constant, contain a number of different traffic types (mix of codecs) and the session hold-time is variable.

To generate RTP traffic within certain minimum and maximum bandwidth thresholds, the IP-to-IP Gateway Test Suite allows users to configure the upper and lower bounds of the bandwidth utilization for each Gigabit Ethernet interface. Once the upper threshold is reached, the application can be configured to generate signaling-only sessions, or pause call generation until the lower bandwidth threshold is achieved.

ToS and DSCP Remapping Verification

IP-to-IP gateways will need to remap ToS/DSCP values for one value on the incoming stream to another on the outgoing stream.

ToS remapping is required to maintain quality of service guarantees especially when media streams cross service providers' network boundaries. The IP-to-IP Gateway Test Suite supports a canned test scenario to test the ToS and DSCP remapping functions of the gateway. Endpoints can be configured to generate and detect particular ToS and DSCP values within the RTP media stream. When an incorrect ToS or DSCP value is received by an endpoint, a detailed call record with very detailed media information is created.

Troubleshooting Capabilities

To aid in troubleshooting, a comprehensive set of statistics, protocol message information and troubleshooting logs captured by the application are available in real-time. All protocol errors observed by the application protocol stack and other detailed protocol information that are required to identify the root cause of the errors are logged.

Integrated Ethereal

The application supports a built-in signaling monitor and integrated Ethereal. Up to eight independent Ethereal monitors (one for each group of endpoints) can be launched. The Ethereal monitors will only display the signaling traffic for the endpoint-group for which it is launched.

Statistics

A comprehensive list of signaling and media statistics are available to aid in the identification of stability, reliability, performance and scalability issues that could arise during the tests. All statistics including QoS measurements and threshold violations are updated in real-time at a one-second interval and are available in tabular and time-based histograms and graphs. The statistics views can be customized and all statistics can be exported to comma-separated files for post analysis.

Three levels of statistics are available to expedite the identification and resolution problem of areas:

— Global Summary Statistics

The global summary view provides a high-level summary of the signaling and media statistics for all endpoint-groups in a single screen. From this view, endpoint-groups with signaling protocol errors and or QoS, MOS, MDI, DTMF, service theft, rogue media and other media related issues can be easily identified.

— Endpoint-Group Statistics

Detailed signaling and media statistics are available for each endpoint-group. The media statistics are collected and displayed on a per media/codec type in a single screen. Up to 15 unique media types can be monitored and analyzed in real-time for bandwidth utilization, packet loss, delay, jitter, MOS, MDI and DTMF. Voice/video and DTMF quality as well as other media related thresholds violation are highlighted per media type.

— Detailed Session Statistics

Call records are provided for each session that fails user-defined thresholds for:

- Path/media verification, ToS/DSCP remapping, theft of service, misrouted packets
- QoS/VQT, MDI, MOS, delay, loss, jitter, R-factor
- DTMF digit verification and inter-digit gaps
- Signaling errors during the session establishment and tear-down phases

Within each call record, detailed information is provided for a comprehensive list of session characteristics such as the calling and called parties, signaling and media start and stop times, negotiated, transmitted and received codecs, QoS (delay, loss and jitter), MOS, R-factor, MDI and DTMF analysis.

Up to ten time-stamped periods during which the user-defined thresholds were not met are captured.

Automation

A TCL command line interface (CLI) is available to automate the key features of the IP-to-IP Gateway Test Suite.

Interface Support

10/100/1000 Ethernet

Transport

— TCP, UDP

Network

— IPv4, IPv6

SIP Signaling Protocol

— RFC 3261, RFC 3262, RFC 3265, RFC 2976, RFC 3515, RFC 311, RFC 3263, RFC 2327, RFC 3264

H.248 Signaling Protocol

— ITU-T H.248.1 v3, ETSI TS 283 018 v.1.1.1, ETSI TS 102 333 v.1.1.2

Audio Codecs

— ITU-T G.711 A-law

— ITU-T G.711 U-law

— ITU-T G.723

— ITU-T G.726

— ITU-T G.729

— ILBC

— AMR wideband and narrowband

— EVRC, EVRC-B

Audio Quality Measurements

— ITU-T G.107 E-model

Video Codecs

— H.263

— H.264

Video Quality Measurements

— RFC 4445 (MDI)

DTMF

— In-band

— RFC 2833

— SIP info and H.248 modify/notify methods

Security Protocols

— Transport layer security – RFC 2246, RFC 2459, RFC 3546 and RFC 3268

— IPSec

— IKEv1

ORDERING INFORMATION

N-01C6592 = SIP Flex software

■ **N-01C6339** = Enhanced RTP generation, monitoring and QoS analysis for SIP

■ **N-01C6405** = TLS security for VoIP signaling protocols

■ **N-01C6593** = SIP IMS enhanced software option

■ **N-01C6597** = Command line interface (CLI) for automating the SIP

■ **N-01C6622** = IKEv1 support for the SIP Flex Test Suite

■ **N-01C6624** = EVRC codec

■ **N-01C6625** = IPv6 support for the SIP Flex Test Suite

■ **N-01C6132** = RTP codec package

■ **N-01C6402** = DTMF generation and analysis

■ **N-01C6403** = H.263 and H.264/MPEG-4 video generation and QoS analysis

■ **N-01C6404** = AMR generation and QoS analysis

Hardware requirements: InterWatch R14 platform or InterWatch M7 platform, a Gigabit Ethernet Interface and a secondary protocol engine version 4.0 (SPEv4).

Platform software requirements: Contact customer service department.

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