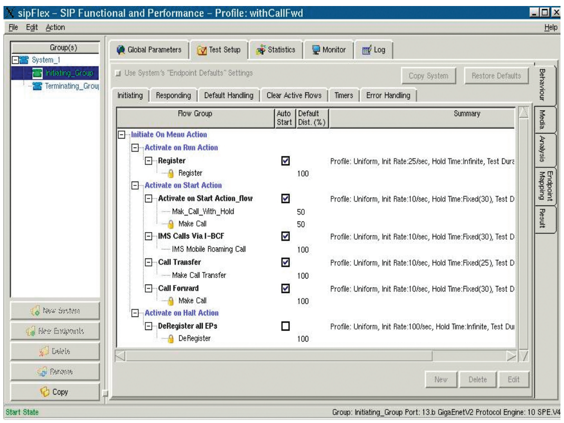
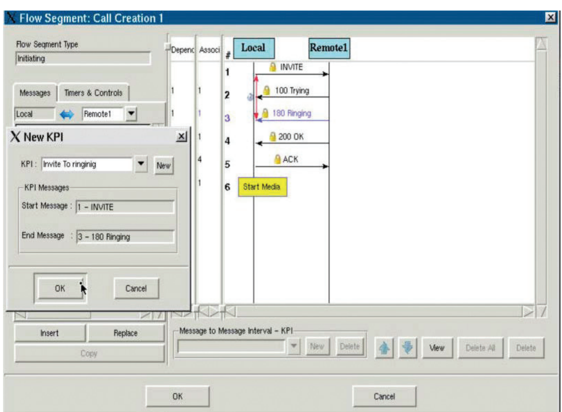
SIP Flex Test Suite

IMS and VoIP Network Element and Service Testing





Highlights

- Feature, negative, load, regression, interoperability and scalability testing
- Negative and proprietary messages and call flow definition
- IMS and Class-5 services simultaneous emulation
- Large-scale and high-performance solution
- IMS AKA with IPSec support
- IKEv1
- Device response latency measurements in real-time
- Comprehensive media test capabilities
- Testing of the following devices:
  - Session border controllers
  - P/I/S-CSCFs
  - Conference servers
  - Presence servers
  - VoIP proxy servers
  - Voice and video mail servers
- Line-rate generation and analysis of RTP streams
- Dual-port Gigabit Ethernet interface
- RJ-45 or GBIC connectors
- Thousands of registrations per second
- Thousands of calls per second (signaling and media)

Next-Generation Network Assessment

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Overview

Next-generation networks are being deployed to enable network operators to provide differentiated services to their customers. These networks consist of several network elements that when interconnected, should provide a reliable and robust network infrastructure, supporting several feature-rich and high-quality services. Developing and deploying these networks present significant challenges to network equipment manufacturers, system integrators and network operators. These challenges include the ability to design and build against specifications that are evolving, get network elements from several vendors to interoperate, verify that the network will be able to support several types of IMS and VoIP services simultaneously and be robust under abnormal and normal network scenarios.

The SIP Flex Test Suite is designed to alleviate all of these challenges with a single, very easy-to-use test application that provides comprehensive test capabilities to enable IMS and VoIP network and service testing during the various stages of development and deployment.

Key Test Features

IMS Network Cloud Routing Emulation

- Emulates IMS network cloud consisting of two or more elements in the IMS network
  Example: UE->PCSCF, UE->PCSCF->ICSCF, UE->PCSCF->ICSCF->SCSCF, UE->PCSCF->ICSCF->SC SCF->IBCF

IMS and VoIP Feature Testing

- Tests Class-5 services at high rates and scale
  Example: Call forward, call transfer, call hold, 3PCC, presence, conference and instant message

- Creates traffic consisting of a mixture of features so as to simulate real-world subscriber behavior
  Example: Generation of thousands of calls per second with the following mix of services: 30 % basic calls, 20 % call hold, 20 % call forward, 10 % three-way conference, 10 % 3PCC, 10 % call transfer – while 50 % of the registered subscribers are subscribing to presence services – publishing, subscribing to and updating presence information

Call Completion Rates and Causes of Failure Simulation

- Simulates call completion rates by simulating user response behavior
  Example: 70 % of calls are responded to normally, 20 % of users are busy, 10 % of users do not answer (keep ringing), 5 % of calls are redirected and 1% of calls are responded to with a user-selected error

Message Floods

- Simulates any SIP message flood consisting of one or more SIP messages
- Simulates registration floods
- Mixes several streams of message floods simultaneously
- Analyzes the impact of message floods on call/service completion rates and network/device response latency

Key Performance Indicators Measurements

- Measures user-defined intervals within each call flow
- Measures network/device response latency under load
- Hardware-based time stamping allows for accurate measurements even under load
- Collects and presents response latency in real-time

Default Protocol Behavior Customization

- Customizable default (spec-defined) protocol behavior
  Example: Handling of re-invites, clearing calls, subscriptions, publications and protocol errors

Protocol Timers Customization

- User-configurable timer values for all protocol timers
- Customizable application behavior once timer has expired
  Example: Do not refresh registration and generate a call once registration has expired

IMS AKA, IPSEC and TLS

- Emulates IMS AKA registrations at high rate
- Establishes IPSec tunnels with P-CSCF
- Uses static or dynamic IPSec
- Customizable IPSec/AKA parameters
- Establishes TLS session for all endpoints

Comprehensive Media Test Capabilities

- Negotiates and transmits several codecs simultaneously
- Negotiates one codec but generate another type with higher bandwidth to test the theft of service protection function of the device under test
- Detects in real-time and at line speed whether the device under test is penalizing RTP streams that do not conform to their negotiated codecs/bandwidth
- Verifies path for every established stream to verify whether:
  - Media was detected
  - Media packets were misrouted
  - RTP codec received was not as negotiated
  - ToS/DSCP value for received packets was not as expected
- Measures QoS for delay, loss, inter arrival jitter and mean opinion score (MOS) with user-defined thresholds
- Provides records for each call that fail the path verification test or exceed the QoS thresholds
- Up to 15 statistics views for 15 combinations of codec, VLAN and ToS values
- User-defined wave files and packetization intervals
- Tests rogue media
- DTMF in SIP info and RFC 2833

Automation and Troubleshooting

- TCL command line interface
- Built-in Ethereal monitor for each Ethernet port
- Detailed call records for user-defined thresholds violation

Real-Time Signaling Statistics

- Provides results in tabular and graphical formats
- Summary and detail statistics per entire system or per group of endpoints
- Signaling statistics per group of endpoints or per flow
- Registrations: successful and unsuccessful registrations with and without authentications
- Calls: successful and unsuccessful calls with and without media
- Session-timer: refreshed, requested, in-progress, successful and unsuccessful
- Event-notification: subscriptions, subscription state and reason code
- Messages: incoming, outgoing and retransmitted
- Errors: incoming and outgoing errors count
- TCP connections: active, attempted, successful, unsuccessful and retransmitted
- TLS connection states: handshake records and errors
- IPSec security associations: active, added, deleted and expired

Real-Time Media Statistics

- Analyzes performance for each individual media/codec type
- Measures packet loss, delay, jitter, R-factor, mean opinion score (MOS) and media delivery index (MDI)
- Validates path of RTP and RTCP packet, detecting teardown time, misrouted, unexpected or multiple codecs
- RTP DTMF and signaling DTMF sessions active or failed and the reason for failure

Performance and Scale per Chassis

- 256 000 unique endpoints/IP addresses
- 128 000 RTP streams
- 256 000 TLS sessions
- 256 000 IPSec sessions
- 256 000 unique MAC addresses
- 256 000 unique default gateways
- Thousands of registrations per second
- Thousands of calls per second (signaling and media)
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Protocol Specifications

Transport

- TCP, UDP

Network

- IPv4, IPv6

SIP

- RFC 3261, RFC 3262, RFC 3265, RFC 2976, RFC 3515, RFC 4028, RFC 3311, RFC 3325, RFC 3891, RFC 3903, RFC 3608, RFC 3428

IMS

- 3GPP TS 24.229
- 3GPP TS 33.203
- 3GPP TS 33.210
- Gm, Mw, Mr, Mg, ISC interfaces

Security Protocols

- TLS
- IPSec
- IKEv1

RTP/RTCP

- RFC 1889, RFC 1890, RFC 2190, RFC 3388, RFC 3551, RFC 3267

Audio/Video

- ITU-T G.711 (PCMU, PCMA)
- ITU-T G.721
- ITU-T G.723
- ITU-T G.726
- ITU-T G.729
- AMR
- AMR-WB
- ILBC
- H.264
- H.263
- EVRC, EVRC-B

DTMF

- RFC 2833

Voice and Video Quality Analysis

- ITU-T G.107 E-model
- ITU-T P.800.1 mean opinion score (MOS)
- RFC 4445 media delivery index (MDI)

ORDERING INFORMATION

N-01C6592 = SIP Flex software

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

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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