

# SIP User Agent

## Functional & Performance Test Tool

### SOLUTION HIGHLIGHTS

Complete Proxy Server and Redirect Server functional and performance testing

Complete Media Gateway Controller functional and performance testing

Emulates up to 32,000 User Agents on each port

Full protocol emulation with standard or customized messages

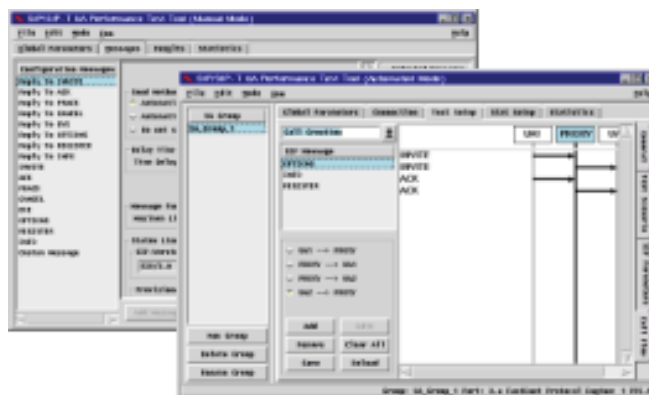
Call flow customization

Support for retransmission and unexpected messages

Flexible Manual Mode for Positive & Negative testing

Extensive set of statistics

Automated testing



### Overview

The Telecom Sector recognizes that scalable next generation IP networks combining voice, data and multimedia not only significantly increase operational savings over circuit-switched or PSTN (Public Switched Telephony Network), but also, provide new revenue streams for business application such as VoIP.

Equipment manufacturers seeking to attract early adopters and a competitive advantage by entering the market for VoIP technology are facing multiple challenges. As standards, features and services evolve, interoperability remains a critical factor.

For Service Providers, integration and deployment of a state-of-the-art multi-vendor network leads to business and technical risks when multiple implementations of VoIP products and standards exist.

Third Generation mobile networks are moving towards SIP to create and manage multimedia sessions through a packet core network. Bypassing the traditional PSTN core network allows a seamless integration between multimedia devices in fixed and mobile networks, but this move makes interoperation and performance in the IP signaling and data planes challenging.

A multi-protocol, multi-featured test system that combines functional, conformance, interoperability and performance testing has proven to be the most efficient way of resolving both equipment manufacturers and service providers issues.

# SIP User Agent

## Functional & Performance Test Tool

### Product Overview

The SIP User Agent Functional & Performance Test Tool for the InterWatch platform emulates multiple User Agents or Media Gateway Controllers and provides users with a flexible manual mode.

Users can generate a high volume of calls through a Proxy Server in order to qualify and quantify performance. Compliant to RFC (Session Initiation Protocol) it can emulate up to 32,000 User Agents/Terminations on each port. Users can therefore validate SIP network devices under extreme signaling load.

In addition to fixed network SIP, the application can optionally be licensed to test the 3GPP version of SIP defined in 24.229. The MGCF, CSCFs and other IMS network components can be stimulated and tested using the SIP application.

The enhanced performance mode allows user-defined call flows and messages to reproduce field conditions. In manual mode when building, sending and receiving messages, the tool's flexibility facilitates positive and negative testing, allowing the user to re-create field conditions and/or interoperability issues.

Combined with the powerful capabilities of the InterWatch Performance & Verification System, the SIP User Agent Functional & Performance Test Tool provides research & development labs of network operators and equipment manufacturers with a complete solution to test next generation converged networks.

## SIP User Agent

### Functional & Performance Test Tool

### Specifications

#### Functional Testing

##### Global Parameters

- Address Configuration
- Source
  - IP Address
  - UDP Port
  - MAC Address or PVC (VPI/VCI) for ATM
  - Domain Name Server
  - IP Address
  - DNS Timeout Period
  - Subnet Mask
  - Outbound Proxy
  - IP Address
  - MTU Size

#### Troubleshooting Mode

Real-time SIP protocol message view

#### Configuration Messages

Reply to INVITE  
 Reply to ACK  
 Reply to PRACK  
 Reply to CANCEL  
 Reply to BYE  
 Reply to OPTIONS  
 Reply to REGISTER  
 Reply to INFO  
 INVITE  
 ACK  
 PRACK  
 CANCEL  
 BYE  
 OPTIONS  
 REGISTER  
 INFO  
 Custom Message

#### Statistics (Incoming/Outgoing)

INVITE  
 ACK  
 PRACK  
 CANCEL  
 BYE  
 OPTIONS  
 REGISTER  
 INFO  
 1xx Responses  
 2xx Responses  
 3xx Responses  
 4xx Responses  
 5xx Responses  
 6xx Responses  
 Unknown

#### Performance Testing

##### Global Parameters

- Topology  
 PVC (VPI/VCI) for ATM  
 MAC Address  
 SIP UA Configuration  
 Phone Number (optional)  
 Username of SIP Address  
 Password  
 Domain of SIP Address  
 IP Address  
 UDP Port  
 Subnet Mask  
 Associated Proxy

#### Associate Proxy Configuration

Domain Name  
 IP Address  
 UDP Port

#### SDP Parameters

Parameter (v,o,s,c,t,a,m)  
 Attribute  
 Value

#### Test Configuration

Endpoint Selection  
 (Local/Remote)  
 Sequential  
 Random  
 Header Format  
 Full  
 Short  
 Call Pattern  
 Receive Only  
 Single Call  
 Uniform  
 Step  
 Burst  
 Call Rate  
 Initial Call Rate (calls/sec)  
 Final Call Rate (calls/sec)  
 Call Rate Duration (infinite - sec)  
 Increment Call Rate (calls/sec)  
 Call Hold Time  
 Infinite Call Hold Time  
 Fixed Call Hold Time (range in sec)  
 Random Call Hold Time (range in sec)  
 Timers  
 T1  
 T2  
 T4  
 TD  
 Ring Timer  
 SIP Response  
 Enable/Disable Reason Phrase

Max Forward Message Header  
 Contact Expiration  
 Enable/Disable  
 Value

#### Call Flow Configuration

Restart Sequence  
 REGISTER  
 OPTIONS  
 Call Creation  
 INVITE  
 ACK  
 OPTIONS  
 INFO  
 REGISTER  
 Call Deletion  
 BYE  
 OPTIONS  
 REGISTER (Expires: 0)

Capability to import manually defined messages into the call flow

Real-time Statistics (Global, Group and Cumulative)  
 Outgoing  
 Number of INVITE (for all endpoints)  
 Number of Successful Calls  
 Number of Unsuccessful Calls  
 Errors and Frequency  
 Incoming  
 INVITE  
 Number of Successful Calls  
 Number of Unsuccessful Calls  
 Errors and Frequency

#### General

Interfaces  
 OC-48c/STM-16  
 OC-12c/STM-4  
 OC-3c/STM-1  
 UTP-5  
 DS3/E3  
 DS1/E1  
 10/100 BASE-T  
 Gigabit Ethernet

#### Network Layer Protocols

ATM  
 IP  
  
 Standard Compliance  
 Session Initiation Protocol, RFC 3261  
 3GPP 24.229

#### Part Numbers

01C5693XXX  
 SIP UA F & P Test Tool

*Navtel delivers high performance, high capacity VoIP testing solutions. Recognized globally for their unrivalled ability to test capacity, conformance, functionality, interworking, performance and reliability, our solutions are the ideal choice for those wanting to rapidly develop and deploy next generation converged network infrastructures or components.*