H.323 Simulation & Testing

Simulation Suite for H.323 Calls and Traffic Generation

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INTRODUCTION

One of the key features of a protocols test tool is the ability to generate phone/video calls or service oriented traffic to simulate conditions that occur in a real network.

This feature, usually called Simulation or more appropriately Calls or Traffic Generation, meets the needs of potential users engaged in:

- Testing the correct functioning of new products before release
- Validating and debugging software
- Load testing during Test & Verification
- Measuring and setting Quality of Service (QoS) and Service Level Agreement (SLA) values
- Interoperability testing between devices and services from different manufacturers and carriers before activating new services
- Verifying and controlling the billing system.

THE SUNRISE TELECOM SOLUTION

Sunrise Telecom has developed a comprehensive set of emulation suites to test different protocols for VoIP, including **H.323**, **SIP**, and **MGCP**. Test Suites are also available for **ISDN**, **SS#7**, **SigTran**, and **Analog**.

This is the application note for the H.323 simulation suite.

The current hardware platforms that support the H.323 Simulation suite are:

- **NeTracker 1000, 3000, and 600**
- **STT-MSA**

H.323 SIMULATION SUITE

The H.323 Simulation suite provides complete coverage of ITU H.323 v.2 and v.4.

It allows simulation of different network elements such as:

- Single and multiple terminals
- Single and multiple gateways

Supported Protocols

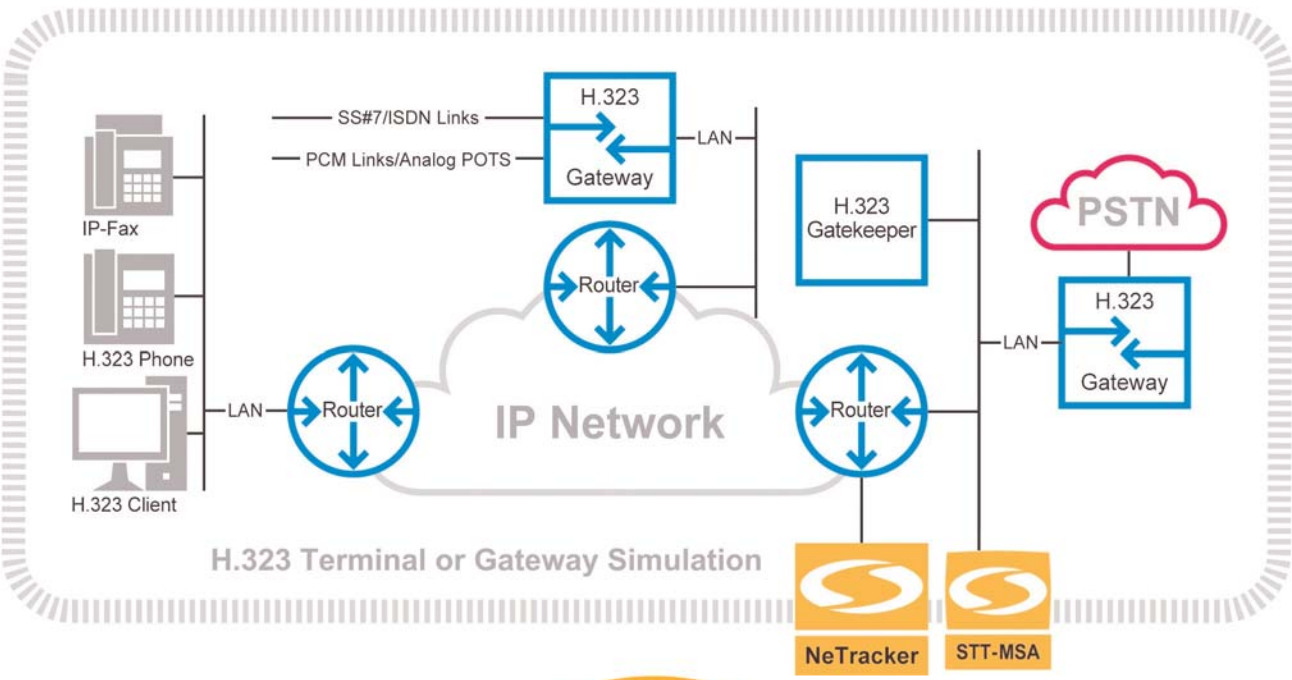
The package supports the following protocols:

- RAS/Q.931
- H.225
- H.235
- H.245
- RTP/RTCP

SIMULATION MODES

The following simulation modes are available:

- **Registration only** for a gatekeeper-stress test
- **Call signaling only** (H.225, Q.931, and RAS) with or without gatekeeper registration; with the "signaling only" mode users can generate voice, video, fax, and data calls.
- **Call signaling with H.245** channel initiation, with or without a fast connect procedure
- **Call signaling with H.235** (Security and Authentication) protocol. This allows Users to test the security part for the H.323 calls and registrations. Custom versions of the Authentication part are available as options.



SUNRISE TELECOM

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H.323 Simulation & Testing

Simulation Suite for H.323 Calls and Traffic Generation

- **Call signaling with RTP**: transmission of pre-recorded voice or video file for path testing
- **Call signaling with MOS** (Mean Opinion Score) measurement based on either **PESQ** (for voice) or EPSNR (for video) algorithms
- **RTP Transmission only** (i.e. without signaling)

INTER-NETWORKING TESTING

In conjunction with other simulation suites such as SS7-ISUP, ISDN, H323, Analog, or SIP, MGCP suites, users can perform simultaneous simulations on the VoIP and PSTN sides, for a complete inter-networking test.

CONFIGURATION

An intuitive drag and drop interface makes test configuration fast and easy for non-expert users. There is no need for scripting or similar programming languages.

Configuration is performed through 3 main steps:

- Set-up of the elements being simulated
- Set-up of the devices involved in the system being tested (such as router-gateways and gatekeepers)
- Traffic Models configuration

Elements to simulate Set-up

The H.323 simulation suite allows simulation of hundreds of terminals or gateways, (depending upon the platform) in any combination, singly or in groups.

Terminal configuration allows single or multiple IP addresses. If a simulated terminal has multiple IP assignments, it becomes a group of terminals.

Gateway simulation allows one IP address to support multiple phone numbers.

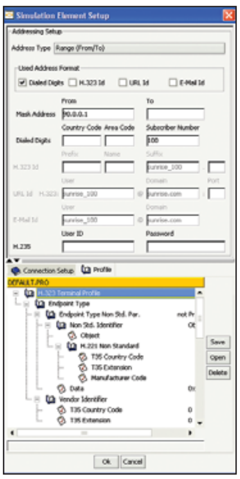
Multiple terminals or gateways may be configured to register themselves to a gatekeeper in order to carry out stress tests.

Phone numbers Set-up

Phone number set-up supports both the ITU E.164 numbering notation and the H.323 Identifier format.

A wild-card system allows the creation of ranges of addresses. For example, if a user needs to simulate 99 E.164 phone numbers with values from 703-403-5000 to 703-403-5099, the entry will be 703-40350??.

Wildcards are also available for alphanumeric fields in the Local Party Number (H.323 numbering format).



Phone numbers and protocol parameters set-up

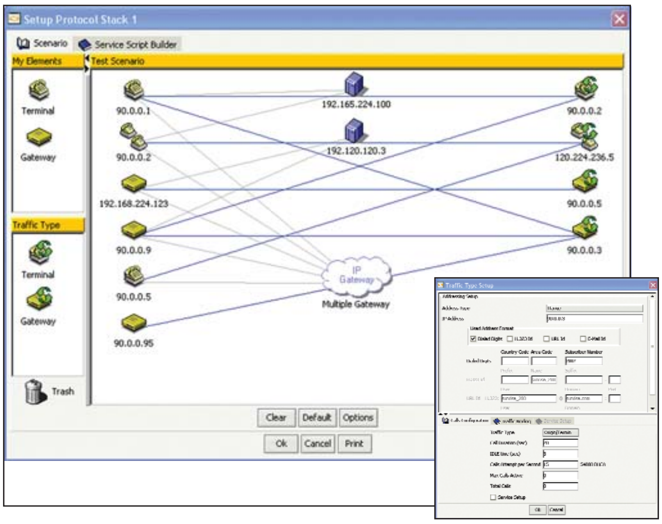
Messages Parameters configuration

Thanks to an easy to use tree-view system users can set all the most significant parameters in the H.225, RAS, Q.931, and H.245 protocols messages and save them in libraries for an easy load and run test.

Gatekeeper and Gateways in the System being tested

Single or multiple gatekeeper connections are possible. Users can configure entities (terminals or gateways) connected to different gatekeepers.

The simulator allows simulation of elements located over different IP sub-networks by supporting multiple router-gateway addressing systems.



Example of a complex scenario

Traffic Models Configuration

Once the simulated entities (Terminals or Gateways) are set, the suite allows users to configure traffic models by setting the following parameters:

- Traffic type: either Terminated or Originated Calls
- Call duration
- Idle time between Calls
- Calls per seconds
- Enable or Disable Service Script function (allowing to generate RTP Traffic)

SENDING VOICE & VIDEO: SERVICE SCRIPT FUNCTION

The Service Script function allows users to generate voice and Video traffic using pre-recorded files or customized audio/video files in RTP packets to simulate real voice conversation.

The same audio/video file can be transmitted several times in the same call.

Audio/video files can be transmitted and received back for path testing.

Voice samples are based on G.711, G.729 or G.723 codecs, while video samples are based on H.263, H.261MPEG-2 and MPEG-4 codecs. DTMF tones can be transmitted and received over RTP/RTCP for testing automated voice services like IVR, Call Centers or Voice Mails.

Jitter and packet loss values can be configured to introduce IP network impairments intentionally during packet transmission and analyze the reaction of the system being tested.

Intrusive QoS (voice & video) measurement

The Service Script allows also intrusive Quality of Service measurements.

Loop-back voice/video calls are generated so that audio/video files are sent and received back.

With the comparison between the source and the received audio files voice quality can be evaluated using either PESQ (for audio) or EPSNR (for video) algorithms, thus providing a QOS score for each call.

Non-Intrusive QoS Measurement

VQmon/SA E-Model algorithm: Sunrise's implementation is based on the well-established ITU G.107 E Model, with extensions to support time-varying network impairments.

It provides call quality metrics, including listening and conversational quality scores and detailed information on the severity and distribution of packet loss and discards due to jitter.

NiQA (Non-intrusive Quality Assessment) algorithm is designed to assess the listening quality of live customer traffic on the basis of the newest **P.563** specification recently released by ITU.

USING SEVERAL TRAFFIC MODELS FOR COMPLEX SCENARIOS

Thanks to the flexible configuration, users can combine different elements to simulate (Terminals and Gateways) and traffic models to create scenarios from simple tests with one call type only, up to complete network simulations with multiple users generating different calls with heterogeneous types of voice traffic.

H.323 SIMULATION RESULTS

When the simulation is running, the H.323 simulation suite provides a unique set of analysis tools that includes:

Trace

Protocol Trace with full message decoding, custom trace, and graphical format (ladder diagram).

Call Detail Record

CDRs are provided for each call (generated/received); CDRs include QoS measurement based on Jitter, latency, packets lost and MOS scores (if either PESQ or PSQM measurement is enabled).

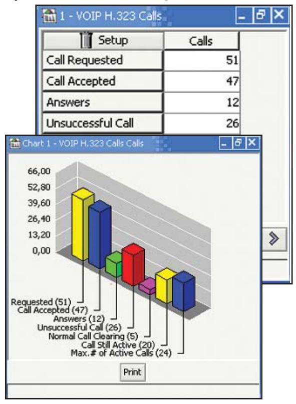
The **CDR-to-frame** function automatically recovers all the related signaling frames and shows the entire call flow in a graphical format (Arrow Diagram); while the **Frame-to-CDR** function automatically recover the CDR related to a selected signaling message.

Audio Playback function is available for each call for which the recording was enabled.

Statistics

A complete set of statistics (in tabular and graphical formats) covers all the main protocol events:

- **Signaling messages**: a dedicated statistic is provided for every signaling protocol stack (H.225 RAS, Q.931, H.245, and H.235)
- **Calls**: a dedicated statistic allows for call completion analysis
- **Disconnection causes**: several dedicated statistics provide the reason why some calls or sessions were not successfully established
- **QoS**: statistics on Jitter and MOS scores are based on programmable thresholds.

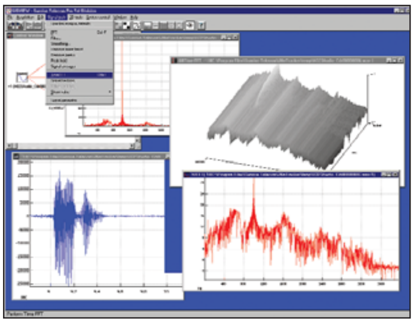


Statistic and chart

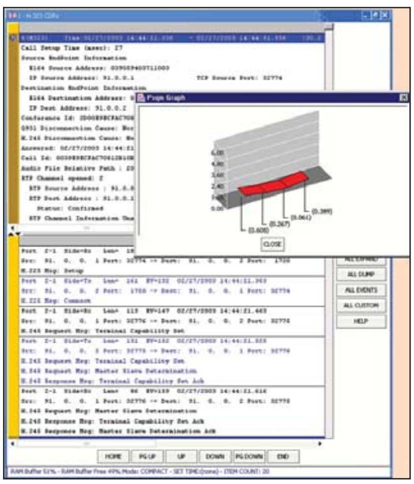
Signal Analyzer

The Signal Analyzer Package allows a post analysis of the audio files recorded over the unit. The package includes:

- a complete signal spectrum analysis
- Signals Comparison functions
- 3D signal representation,
- a scripting language for programming analysis



Audio signal analysis



Example of a CDR and its frames

ANALYSIS TOOLS: FILTERS AND TRIGGERS

Filters are available for both Trace and CDRs. Any number of filters with multiple conditions can be set.

Triggers allow the creation of an advanced traps system that combine protocol events, counters, and timers. Triggers are based on a Graphical Editor.

APPLICATIONS

The H.323 Simulation suite has several potential applications for product designers, manufacturers, and service providers. The main applications include:

- Media gateway testing
- IP Terminal and IP-PBX testing
- Gatekeeper load testing
- Product qualification
- Interoperability testing
- In service QoS

PERFORMANCE

Please contact your local Distributor or Sales Representative to have detailed information on performances.

ORDERING INFORMATION

Suites and Protocol packages:

NeTracker part number	STT-MSA part number	Details
NT-H323-TT	**MSA-H323-TT**	H.323 Simulator up to 0.5Millions Calls per hour with RTP Traffic Generation. It includes the service script environment to create interactive tones and audio files send and receive sequences
NT-H450-TT	**NT-H450-TT**	Implement the Supplementary Services emulation based on H.450 ITU specs over the H.323 protocols. NT-H323-SIM Package is required
NT-VoIP-TT	**MSA-VoIP-TT**	Complete VoIP Simulation Package includes H323, SIP, MGCP, and RTP simulators

Options:

NeTracker part number	STT-MSA part number	Details
OPT-PESQ	**OPT-PESQ**	Intrusive QoS Analysis based on ITU P.862 (PESQ) algorithm
OPT-VIDEO-MON	**OPT-VIDEO-MON**	It enable the video recording and playback
OPT-VIDEO-SIM	**OPT-VIDEO-SIM**	Video over IP simulation suite
OPT-EPSNR-SIM	**OPT-EPSNR-SIM**	Intrusive EPSNR
OPT-SIG-AN	**OPT-SIG-AN**	Signal Analyzer Package for post analysis of the audio files recorded over the unit

All other suites, protocol packages, options and accessories:

Please contact your local Distributor or Sales Representative.

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