SAGE INSTRUMENTS, 240 AIRPORT BLVD., FREEDOM, CA 95019-2614 Ph (831) 761-1000 Fax (831) 761-2452 www.sageinst.com

Release Notes

v5.11.9 for 960x Multi-Channel Test Unit

This document describes a new software release (v5.11.9) for the 960x. This software supersedes Version 5.7.8 and is available now.

NEW FEATURES

1. **Analog Interface Support** — Adds support for metallic analog 2-wire interfaces.

2. **One-Way Delay Test** — This new test supports one-way delay measurements between any two test ports on the 960x. This test is only available as part of the Next Gen test package (SMOS, PVIT, Echo Sounder, Echo Generator Test Lines and Directors, One-Way Delay Test)

3. **MF/DTMF Digit Analyzer** — This is a 'smart' digit analyzer, that can automatically determine whether received digits are MF or DTMF, and report them accordingly. Also, a user can specify four thresholds for qualifying a digit: Minimum On-duration, Minimum level of each tone, maximum level difference between the two tones of a digit, and maximum frequency error of the tones in a digit.

4. **MF/DTMF Digit Sender New Features**
   a. Independently adjustable digit on and off durations
   b. Independently adjustable level for each of the tones making up a digit
   c. Independently adjustable frequency deviation (in Hz) for each of the tones making up a digit
   d. User can optionally use a dial pad to ease digit entry

5. **ITU-T G.107 E-model Test** — The E-Model test is based on ITU-T Recommendation G.107. It estimates end-to-end voice quality, taking IP Telephony parameters and impairments into account. 960-measured parameters are used as inputs to the ITU-T G.107 E-model to produce a transmission rating factor R value, and the derivative GoB (Good or Better), PoW(Poor or Worse) and E-MOS (E-model-based mean-opinion-score) numbers.

   For the TDM interfaces (T1/E1) and analog interfaces, the E-model test is included as part of the IP telephony transparency test (or MoIP test).

   On the Ethernet interface, the E-model values are always provided as long as an RTP (Real-Time Transport Protocol) stream is received by the 960B. Currently, the SIP (Session Initiation Protocol) Call and RTP monitoring tests will report E-model results.

6. **Echo Sounder and Echo Generator** — These two functions are now available for SIP calls.

7. **ICMP (Internet Control Message Protocol) Ping Server** — A 'ping server' can now be provisioned for one IP interface at a time. It will answer a 'ping' command from a far end location, so is helpful in confirming IP connectivity to a 960B IP port.

8. **SIP Call Setup** —
   a. Now allows a user to specify the ToS (type of service/quality of service) level for the SIP call data stream.
   b. Also supports username/password authentication, and user adjustable jitter buffer size.
   c. Responder functions will now automatically "register" with a proxy call server upon activation.

9. **SIP Monitor** — This function can be configured to capture data for a specific IP address, or all addresses. It can be very useful for analyzing SIP call-setup problems. The 960B will support one SIP Monitor function running per ethernet port.

10. **Audio Monitoring of IP Calls** — This new function allows audio monitoring of IP calls, in much the same way as has been supported for PCM calls.

11. **IP Full Duplex Monitor Mode** — This test access mode is for use with an 'ethernet monitoring tap' device. When used with such a device, you can passively monitor both directions of transmission. You can use the SIP Monitor or RTP Monitor functions.

12. **One Way Delay Rcv and DS0/Fractional BERT added to PCM Monitor Mode** — The One Way Delay receive function and DS0/Fractional BERT have been added as available choices for channels within a PCM interface which has been configured for Monitor mode.

ENHANCEMENTS

1. **INMD Test Now Captures the Echo Sample When Logging Enabled** — When logging is enabled for the INMD (*In-Service Non-Intrusive Measurement Device*) test, the 960x will now automatically capture 5-second samples of the source and echo audio signals.

2. **RTP Monitor Now Reports Packet Statistics** — Now displays number of Received Packets, Lost Packets, Delayed/Discarded Packets, Out of Order Packets, and packet Inter-Arrival Jitter.

3. **Span View Enhancements** —
   a. **Can Now Be Sorted** — In the Span View, you can now sort the displayed lines by clicking on the desired column title button. You can also toggle between ascending and descending order by clicking the column title button repeatedly.

   b. **Provides More Channel Status Information** — The new display format is organized in five (5) columns. From left to right: Channel Number, Test Type, Destination, Log, and Status.

      The Channel Number column is the same as previous software versions, and is self explanatory. The Test Type column displays the test type defined for that channel. If no test type is defined, displays "Idle". The Destination column: For test Directors, displays the target far end phone number or target IP address. For test Responders, displays "Responder". The Log column indicates, for each channel, whether logging is enabled. The Status column displays various call states and test states as they progress.

4. **Improved SMOS Robustness** — Previously, if there were significant dropped voice packets while the Responder function was transmitting its answer tone, the Director mistakenly assumed the answer tone was completed and launched into the test. In such instances, the responder ignored Director commands because the Responder was still sending the answer tone. Now, the Director function ignores answer tone gaps up to 300 milliseconds.

5. **Test Results Logs Now Contain Destination Phone Numbers** — When tests are run on multiple DS0 channels, and all are logged to the same log file, it can be impossible to determine which test execution called what destination phone number. Now you know.

6. **Send Tone is Gone, Use Send/Measure Tone** — The Send Tone function has been eliminated. You should now use the Send/Measure Tone function. This function allows you to simply measure tone, or send a tone while you are measuring.

CORRECTIONS

1. **On 960x-originated SIP Calls, the Responder Function Will Automatically Hang up** — 960x Director functions now issue a BYE message when ending a test call. When a responder test function receives the BYE command, it will automatically hang up, reset, and wait for the next call.

2. **Unused Echo Generator Settings Are Now Disabled and Greyed out** — When a user selects Loopback, Packet Loss, or some other advanced Echo Generator settings, various user settable parameters are not used by the system. They are now disabled and greyed out.

KNOWN DEFECTS

1. For IP calls via a Proxy Server (in the Span/Channel View), the Destination column displays the IP address of the Proxy Server instead of the target IP phone.

2. When a PCM port is configured for ISDN emulation, the D-Channel message log may contain occasional erroneous data./messages.