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Release Notes

v6.4.12 for 96x Multi-Channel Test Unit

This document describes a new software release (v6.4.12) for the 96x. This software supersedes Version 5.11.9 and is available now.

NEW FEATURES

1. **PCM Capture Now Supported in Analog & IP Tests** — The PCM Capture capability has been extended to tests running on Analog and IP (Ethernet) test access ports.

2. **PCM Capture Setup Now Supports a "Trigger" Setting** — The PCM Capture setup screen now has a "Trigger" button. Clicking on that button opens a setup screen that allows you to do the following:
   a. Enable or disable the Trigger mode
   b. Set the "Trigger Peak Level" (the threshold level setting which starts PCM Capture). You can specify a trigger level ranging from -3 dBm to -30 dBm.
   c. Specify a "Pre-Trigger Time", which allows you to start PCM Capture just before the Trigger threshold is exceeded. You can specify a lead time of 0 ms to 1000 ms (0 to 1 second).

3. **PCM Capture Now Supports Synchronized (Time-Aligned) Capture of Two Audio Streams** — This capability is available in terminate or single-monitor modes. The PCM Capture screen now has a "Sync" check-box to enable the function. Such synchronized PCM Capture can be quite useful when the relative timing of the two PCM audio streams need to be preserved for analyzing echo, modem, and director/responder activity.

4. **MOIP (Modem Over IP) Test is Now Supported on Analog Test Access Ports** — uses the same MOIP setup screens as found when using PCM test access ports.

5. **RTP Monitor Now Has a "No Packet" Timeout Feature** — Audio packets can halt when the talker's "Hold" button is pressed, or when both parties stop talking and silence suppression is engaged on the circuit.

   Now, when the audio packet stream has stopped for the specified timeout period, the 96x will disengage from the RTP stream. The RTP Monitor configuration screen now has a "No Packet Timeout" window where a user can enter a timeout value ranging from 1 to 999 seconds.

6. **PVIT Test is Now Supported on Analog Test Access Ports** — uses the same PVIT setup screens as found when using PCM test access ports.

7. **Analog Test Access Now Supports 4-Wire Mode** — The 4-wire mode is only supported on Analog test access ports #2 and #4 (These ports have two modular jacks: RJ11 for 2-wire mode, and the smaller RJ22 for 4-wire mode)

   Two forms of 4-wire access are provided:
   a. "4-Wire, Normal" — This is a "dry circuit" mode, with no DC potentials applied by the 96x
   b. "4-Wire, Cellphone" — No DC potentials are applied by the 96x, but there is a "sense" resistor applied across the microphone pair of leads. This resistor fools the cell phone into thinking there is a headset attached, thereby muting its built in microphone and earphone. Also, one lead of each pair is tied together to form a "common" lead for interfacing to the cell phone headset jack.

The 4-wire interfaces are intended only to connect to a telephone handset interface or cell phone hands free headset jack. Note that the 4-wire interface is therefore NOT a calibrated 600 ohm interface, so any measured levels are for indication only, and are not to be assumed "accurate".

When you double-click on the analog "channel" in the "Span/Channel View", the top right of the resulting screen has two level adjusting up/down scroll boxes; one for Tx gain and the other for Rx gain. These controls allow a user to compensate for the varying interface characteristics of various phones and cell phones.

Sage Engineering suggests that a good starting point is to set the Tx gain for "-12", and the Rx gain for "0".

8. **RFC 2833 Monitor** — Is a new function selectable under the RTP Monitor test. This function can report the parameters of DTMF and MF Digits, and various Telephony Tones and Signals such as dial tone, ringing, busy tone, etc., IF they have been encoded in special RTP payload formats.

   The events are time stamped. Digit/tone parameters include level in dBm, duration, high frequency, low frequency, and modulation frequency if present.

   For each event, this function also reports the RFC 1890 static "payload type".

9. **One-Way MOS Test** — Because some devices were designed with simplistic data detection algorithms, they may erroneously switch to data mode when encountering the standard SMOS director/responder test. This new one-way test will avoid such problems.

   In the test configuration screen, this test also allows you to specify that results be *averaged* over a user-specified time interval, then periodically logged/plotted at the end of each interval. The user may specify a time interval ranging from 1 second to 9999 seconds.

10. **Command Line Interface** — A basic command line interface (CLI) is now included in units equipped with the Ethernet remote control option (Option 800). When toggled to Command Line mode, the 96x supports up to 10 simultaneous users. When toggled to GUI mode, it reverts to single user mode. The 960B command line mode shares many CLI commands with the Sage 966R Remote Digital Test Server.

ENHANCEMENTS

1. **FAX Send & FAX Receive Functions Now Time Stamp all Events** — In addition to time stamping each fax protocol event based on the user's PC clock, the 96x now gives DSP *start* and *end* "tic" times of each event. This allows precise calculation of the event's signal duration.

   This enhancement is available for the PCM and Analog test access interfaces.

2. **ISDN Call Configuration Screen Now Allows User to Specify B-Channel Capability On a Per Channel Basis** — A user can specify Speech (default), Unrestricted Digital, Restricted Digital, 3.1 kHz Audio, Unrestricted w/Tones, Video, 56K Data Call, and 64K Data Call.

3. **DS0/Fractional BERT Configuration Screen Now Has a "Select All" Button to Simplify Full-Rate PCM Tests** — You no longer have to click 23 other channels to provision a full-rate BER test.

4. **Manual Call Control Now Available for Analog Calls** — A user can now establish a call manually, and change the test without hanging up the line.

5. **For SIP Calls, "E-model" Tab, Now Also Displays and Logs Current Values Plus Historical Min & Max Values**

6. **For SIP Calls, Now Allows User to Graph the Average R-value, GOB%, POW%, and E-MOS Value over a User-specified Interval Ranging from 1 Second to 100 Seconds per Plot Point**

CORRECTIONS

1. **In Windows 98, the "Home" Screen "Calls Attmpt" Column Is Not Always Updated** — This no longer happens.

2. **Loading a Test Configuration File While in the IP Channel View Can Cause the 960 GUI Program to Crash** — This no longer happens.

3. **SIP Call Configuration Does Not Remember Changes** — When in the SIP call configuration screen, go to the "Make Call" tab and enter an IP address in the "DN Addr" field and enter a port number in the "Port" field, then click on a different tab. Now click the "Make Call" tab – the data you entered has reverted to their previous values. Similarly this also happened if you check-marked "Via Proxy Server" and entered the Proxy Server and Port number.

   This no longer happens.

4. **At the 960 GUI "Home" Page, the "Calls Failed" Counters for Analog and Ethernet Interfaces Are Not Operational** — The counters are now operational.

5. **In the PCM Configuration Screen, the Term "Pulse Shaper" Is Not Familiar to Telephony Personnel** — This label has been changed to the more familiar term "Line Build-Out".

6. **In a SIP Call, the Results Under the "Details" and "E-model" Tabs Disappear When the Test Completes** — The results now remain under these tabs after the test ends.

7. **On SIP Calls, the Packet Loss Counter Result Is Not Logged in the General Results File** — Now it is.

8. **On SIP Calls, the Sip Log (*.csv format) Mixes Some Column Header Information in with the Values, Causing Problem Importing to Microsoft Excel** — Fixed.

9. **On Analog calls, the Reported Dial Tone Delay is 16 ms until the actual DTD is over 1045 ms** — Fixed

10. **On the Analog Interface, Ring Voltage Measurement Is Not Accurate** — Fixed

11. **One-Way Delay reports "DSP Test Died.."** — When One-Way Delay TX is used for the Director function, and RX is used for the Responder function, the One-Way Delay TX function stops at the end of the first test, with the error message: "Error: DSP Test Died (Call Up)". This has been fixed.

12. **Loops up T1 CSU, but Doesn't Loop it down** — Fixed.

KNOWN DEFECTS

1. **When PCM Capture Is Activated with the SYNC Option Check-Marked, PCM2 or PCM4 Will Not Capture Data** — This is fixed in unreleased "Engineering Test" T1 DSP files v65.4 or later. Contact Sage Technical Support if you require this minimally tested patch.

2. **The 96x May Reject a SIP Call Message If the Header Is Not All Uppercase** — This can result in problems involving SIP call registration. This is fixed in unreleased "Engineering Test" IP DSP files v47.10 or later. Contact Sage Technical Support if you require this minimally tested patch.