Voice Over Packet Test Applications

An Ameritec White Paper

A white paper describing the use of GoldenVoice™ to measure Signal-to-Noise Ratio, isolate non-signal energy and confirm call routing in a packet-switched environment

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VoP Test Applications

Using GoldenVoice™ to measure Signal-to-Noise Ratio, isolate non-signal energy and confirm call routing in a Packet-switched environment

Contents

VoP^3 Test Applications

I. Introduction

This document is the third in a series of white papers developed by Ameritec Corporation to assist the development community in the testing and isolation of problems in today's packet-switched network environment. The prior documents, "*Voice over Packet Application Guide,*" published in October 1998 and *"The VoID in VoIP Testing"* published in August 2000, addressed how specific measurements are made and how the measured impairments relate to problems in the various components of a packet-switched network.

The intent of this document is to focus primarily on the new capability associated with the additions of the VoP^3 feature set to Ameritec's voice over packet (VoP) test solution. VoP^3 consists of three added features: (1) the ability to measure the signal-to-noise ratio on all circuits (channels) concurrently using GoldenVoice as a stimulus; (2) the ability to isolate within specific frequency bins non-signal energy that occurs above a user-defined threshold; and (3) the ability to use multiple GoldenVoice patterns to perform path checks across packet networks.

A. History

In 1998 Ameritec Corporation introduced its first call generator-based solution for testing the effects of transporting voice over a packet-switched (VoP) network. The market asked for solutions that would allow it to measure circuit impairments, such as voice delay and packet loss, on subscriber-to-subscriber connections. Ameritec's initial product offering specifically addressed these needs and for the first time gave developers the insight into the nature of the impairments based on quantative measurements of an end-to-end call session. Because a call generator was used as the basis for the testing, developers could see how various load conditions impacted the subscriber experience.

As the technology evolved, so did the requirements for test equipment and in early 2000 Ameritec introduced VoP^2, Ameritec's second generation offering that provided additional capability and precision. The market knew that the network was not symmetrical and required a method of precisely measuring one-way delay and an improved precision for measuring round trip delay in both single and multi-switch applications. Additional requirements were the ability to count lost voice packets and to measure packet loss duration; measure front-end and back-end clipping and measure voice packet jitter. These features were the basis for Ameritec's VoP^2 product offering. As described in Ameritec's white paper, "*The VoID in VoIP Testing*," the VoP^2 tool gave developers and validation engineers the quantative data necessary to isolate problems within the various components of a packet-switched network. VoP^2 also offered the ability to test single interfaces in a single gateway, and for the first time allowed interworking testing of various physical interfaces locally or in a packet-switched network deployed over a wide area.

As packet-based voice applications move from the enterprise environment to the public network the requirement to achieve circuit-switched network quality becomes a more significant development objective for packet-switched network equipment manufacturers. The ability to confirm routing through the network, measure signal-to-noise ratio (SNR) and isolate non-signal energy grows in importance. This is a significant challenge because traditional single tone stimulus for measuring SNR and confirming path routing will not function in a packet-switched network. Ameritec has solved these technology challenges in its new VoP^3 product offering by increasing the number and capability of unique GoldenVoice patterns available for use as a stimulus in these demanding tests.

B. Uniqueness of Ameritec's VoP^3 solution

The VoP equipment manufacturers have been challenged to improve the performance and voice quality of their products in order for VoP technology to be adopted as the standard infrastructure for transporting voice calls in the public network. Consequently, developers and engineers must be able to isolate faults, make changes and then validate that their corrections and enhancements have a positive impact on the performance of their product.

Service providers are also challenged. They must integrate dissimilar elements from different manufacturers into their existing network and ensure interoperability. They must also ensure real-time network performance meets standards for quality of service. The dynamic nature and complex topology of the VoP network complicates these challenges. Voice quality is adversely impacted by many different elements of the network. It also changes on a call-by-call basis as network load conditions and resources vary. The end result is that it is difficult for manufacturers and service providers to isolate and correct voice quality problems with existing test tools. More detailed information is needed.

This type of testing requires the measurement capabilities found only in Ameritec call generators: (1) the availability of a stimulus, GoldenVoice, that can be used for all VoP measurements (delay, jitter, clipping, dropouts, SNR and path confirmation) regardless of network type; (2) the ability to execute VoP testing on one to thousands of channels simultaneously; (3) results that are based on objective measurements; and (4) the ability to configure each channel to simulate the characteristics of either the calling or the called party.

II. What is GoldenVoice?

Ameritec is a worldwide leader in call generation and analog transmission impairment (TIMS) test products. TIMS products use single or complex tone sequences as a test stimulus and measure impairments to this signal caused by the network. Some typical parameters that can be measured in this manner are signal attenuation, noise, signal-to-noise ratio, delay and signal dropouts. This type of testing can very precisely characterize the quality of a transmission path.

Unfortunately, many of today's commonly used codecs and vocoders (G.711, G.723.1, G.726, G.728, G.729A/B, etc.) use voice compression algorithms that do not reliably pass the single or multi-frequency tones needed to perform this type of testing . Ameritec developed GoldenVoice in response to the requirement for a test stimulus that would pass through these codecs. It possesses a dynamic range and spectral content that approximates human speech. Codecs cannot discern GoldenVoice from real speech and pass the signal through, thus making traditional impairment measurements possible.

III. New VoP^3 Capabilities

As mentioned earlier in this document VoP^3 provides an additional set of measurements beyond those of the VoP^2 product including:

A. Signal-to-Noise Ratio (SNR)

The signal-to-noise ratio (SNR) is a key measurement used to assess the performance of traditional voice and data networks. It provides a means of directly comparing the amount of desired signal energy to undesired noise energy present in a transmission path. It is considered to be a true indication of the quality of an audio signal. Although one would think that an insignificant amount of noise would exist in a packet network, in reality there are many potential sources of noise which can degrade the signal quality, such as codecs, vocoders, transcoders, A/D and D/A converters and other network elements.

Ameritec's VOP3 solution provides more than just a SNR value. It also measures the amount of signal energy, spurious energy and total energy received.

Signal-to-Noise Ratio:	Average and maximum SNR received (from 0 to 39dB).
Signal Energy:	Average and maximum GoldenVoice energy received (from 0 to -50dBm).
Spurious Energy:	Maximum non-GoldenVoice energy received (from 0 to –50dBm).
Total Energy:	Average and maximum GoldenVoice energy plus extraneous noise received (from 0 to –50dBm).

Noise Isolation

Obviously, a low SNR value can be attributed to either a low signal level or a high noise level. The VOP^3 capability's SNR test can provide a development engineer with the necessary data to determine which is the problem condition. However, a problem arises if the cause of the low SNR is a high noise value. How can this noise be characterized to the extent that its cause can be isolated? VOP^3 provides an additional diagnostic test that provides this characterization. This test allows the user to enter a threshold above which spurious noise will be detected. When the noise exceeds this threshold, it is analyzed and the frequency component with the highest level is reported.

Spurious Noise Detected:	Noise is analyzed and the frequency component with the highest level is reported into one of 24 bins centered on frequencies in the range of 160Hz to 3840Hz.
Failed to Detect Energy:	Register increments if energy is greater than 0dBm or less than –50dBm. Provides an indication of extremely high

	noise levels or loss of consecutive voice packets.
Total Energy:	Average and maximum GoldenVoice energy plus extraneous noise received from 0 to –50dBm.

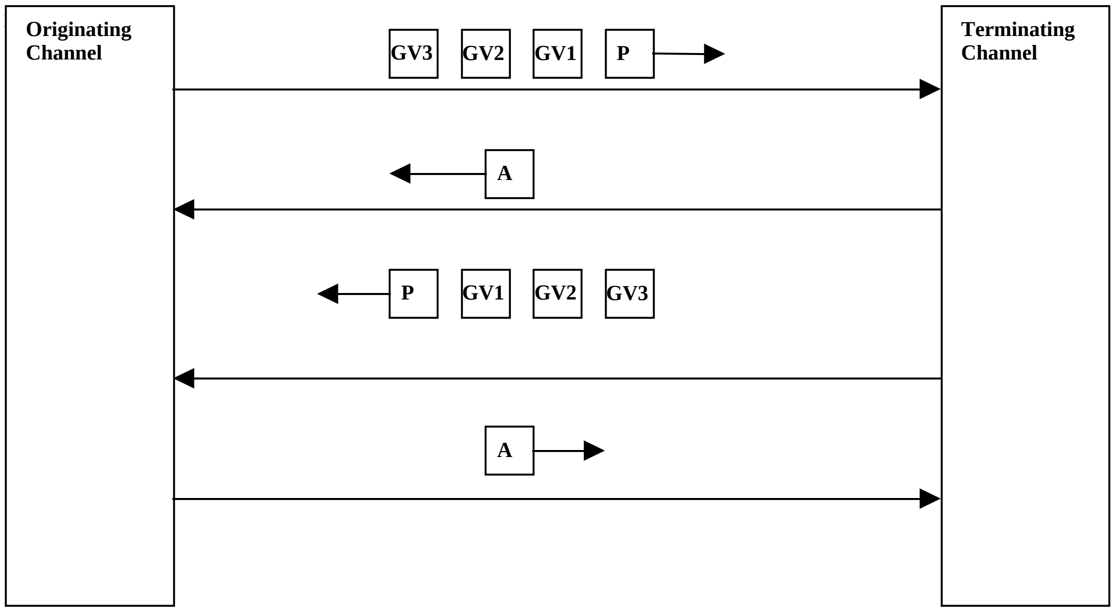
B. Voice Path and Routing Confirmation

A critical test that is traditionally performed on switching equipment and switched networks is voice path validation. In this test, a bulk call simulator originates and terminates calls that are routed by the switching system or network. Once the call is answered by the call simulator, a path confirmation sequence is exchanged continuously between each calling and called line for the purpose of verifying the existence of the 2-way voice path. This confirmation sequence can be comprised of one or multiple single-frequency tones; dual-frequency tones, such as DTMF digits; or BERT patterns. This test provides a means of ensuring that every call has a valid two-way voice path present for the entire call duration.

Another typically performed, and equally important, test for switching systems and networks is the confirmation of call routing. In this test, a bulk call generator is again used to make and answer calls. Multiple single-frequency tones are used as the method of path confirmation, but for this test, a unique tone pattern is programmed for each originate/terminate pair. Each pair checks to ensure that it receives the proper tone pattern; if not, the failing side reports an error. Since each pair is programmed with a unique path identification sequence, an incorrectly received tone pattern indicates a routing error.

It has been previously mentioned that codecs utilizing voice compression schemes do not reliably pass the tones traditionally used to perform these important tests. Therefore, until now, these voice path and routing confirmation tests could not be performed in systems other than G.711 (PCM).

Ameritec has solved this obstacle by creating ten unique GoldenVoice path confirmation patterns to be used in place of the commonly used single-frequency tones. The confirmation sequence consists of a pilot tone pattern, 3 identification tone patterns and an acknowledgement tone pattern as diagrammed below.



P = Pilot GoldenVoice Pattern
GV1 = First GoldenVoice Pattern
GV2 = Second GoldenVoice Pattern
GV3 = Third GoldenVoice Pattern
A = Acknowledgement GoldenVoice Pattern

Ameritec has solved a second technical issue necessary for implementing these tests. That issue is the reliable detection of complex tone sequences in the presence of inevitable delay and packet loss. Ameritec call generators can successfully and reliably confirm voice paths in an environment that may have up to 400ms of directional delay, up to 40ms of continuous voice packet loss and signal distortion of up to 10%.

IV. Summary

Ameritec call generators have been the industry standard for verifying the performance of telephone equipment and networks for over twenty years. These products are designed to a high standard of accuracy: zero failures in one million call attempts. They are well known for their exceptional reliability and performance. Ameritec call generators support most commonly used PSTN to VOP interfaces.

- POTS
- T1/E1 CAS
- T1/E1 ISDN-PRI
- T1/E1 SS7
- DS3

Ameritec continues to add industry-leading measurement capabilities to its renowned and widely used VOP test offering. These VOP measurements can be performed locally within a single system or globally between distant endpoints of an entire network via a GPS synchronization feature. The latest release, VOP^3, adds the ability to use time-tested methods of verifying transmission and switching system performance on VOP equipment and networks. A complete suite of VOP measurements ensures a thorough characterization of system performance.

- 1-way Delay
- Round-trip Delay
- Voice Packet Dropouts: Count and Duration
- Clipping: Leading and Trailing Edge
- Jitter
- Signal-to-Noise Ratio
- Spurious Noise Isolation
- Voice Path Verification
- Routing Verification

With the addition of these new test capabilities, Ameritec call generators provide an unmatched suite of measurements to be used for quantifying the performance of and isolating faults in VOP systems and networks.

You can count on Ameritec to provide the tools, the resources and the support to make you a success. If you have questions on testing your application contact Ameritec Support at askzeke@ameritec.com.

