

Sample Telephone Engineering Analysis

IVR Systemic Inconsistencies

Updated 3/2017

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analysis procedures



Overview

Sample Company utilizes an Interactive Voice Response (IVR) system which responds to both voice and DTMF tones. The system administrator reports that the IVR does not reliably respond to either voice or DTMF

John Doe at Sample Company stated that the VoIP switch had already been configured to boost the volume levels by 6 dB, the maximum allowed within the switch. This change was made at the suggestion by the telephone service provider (Telco).

Palitto Consulting Services, Inc. (PCS) was engaged to perform an independent analysis. On September 25, 2013, tests were conducted using DTMF and voice. We looked at potential network issues, and captured digital audio for analysis.

Four additional tests originating from cellular telephones were conducted on September 27.

At the request of the client, additional testing on was conducted on October 16, focusing on response from analog telephone lines.

The details of these efforts, and our conclusions, are presented below.

Key Findings

1. TCP packet analysis showed no problems with packet loss or jitter.
2. No problems with background noise or digital artifacts were noted in any of the signals.
3. The DTMF tones were successful on all tries. We could not cause a failure scenario with DTMF.
4. Voice levels, compared to DTMF, are extremely low
5. The first few milliseconds of voice signals are clipped. For example, if the caller speaks "Two", the consonant is lost and the resulting audio going into the switch is "ooh".

Network Diagram

From the Telco interface, telephone signals enter a VoIP switch, capable of processing both PRI and analog lines. The switch separates the PRI channels and distributes Voice Over Internet Protocol (VOIP) as TCP packets to the ShoreTel server, and to the IVR server, as seen in this diagram.



TCP packets are placed on the internal TCP network, and can be sampled at any point within the network. Digital audio can be intercepted and captured from the switch in PCM format. The switch offers two points-of-capture: before and after internal signal processing (DSP).

The PCS team performed TCP packet capture on the network, and audio capture at both the Pre- and Post-DSP points from the switch.

Test Procedures

The TCP packet stream was captured and analyzed using WireShark.

The VoIP switch was configured to capture raw audio files from the PRI. A series of 60 second recordings were initiated. Each recording captured audio in four ways:

- Raw received audio
- Raw Transmitted audio
- Processed received audio (after VoIP signal processing had acted on the file)
- Processed transmitted audio

The October tests were conducted using analog phone lines connected to a VoIP switch.

The audio files were retrieved and analyzed using Audacity 2.0.3

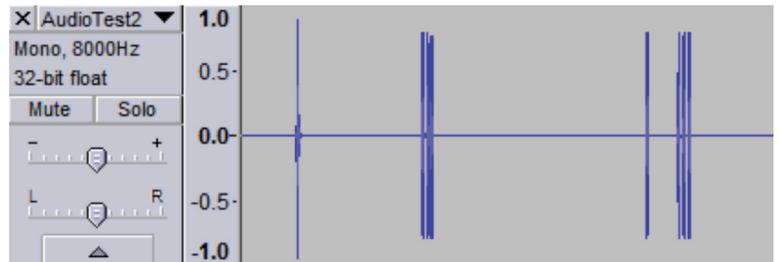
No client data was accessed for this test.

Detailed Analysis

Tests on 9/25/2013 were conducted using a VoIP telephone. Calls were made externally through the Publicly Switched Telephone Network (PTSN) into the client office. All three tests of DTMF-only access were successful.

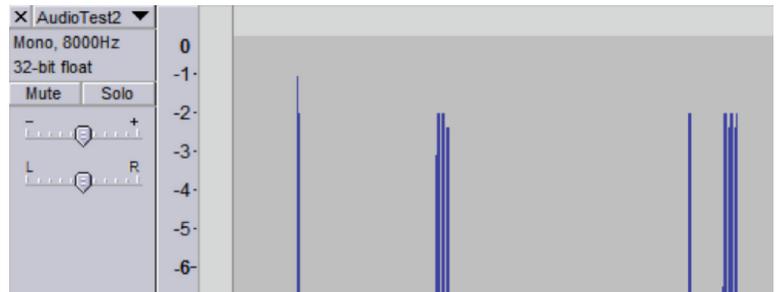
This screen capture from Audacity shows a series of DTMF responses to the IVR prompts

AudioTest20130925.04_raw_rx



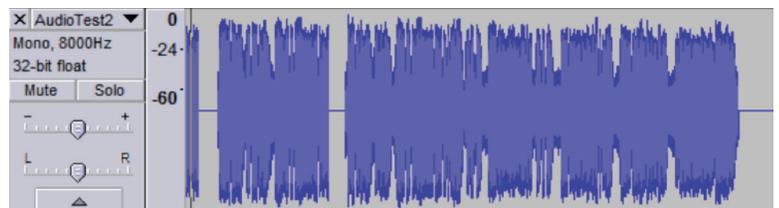
Peak level for DTMF was consistently above -2 dB on all tests

AudioTest20130925.04_raw_rx



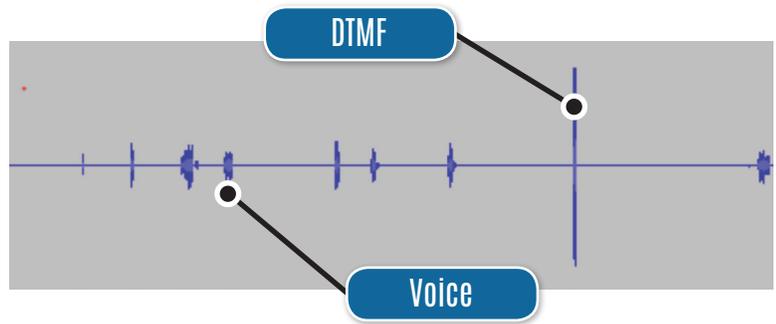
Transmitted audio from the IVR shows average peaks at around -12 dB

AudioTest20130925.05_raw_tx



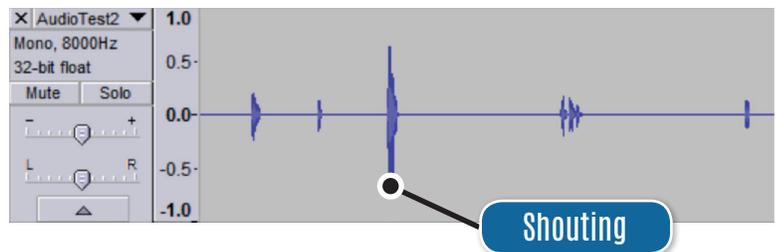
Audio tests demonstrate received voice audio levels average peaks at -21 dB. Compare this to the DTMF tone at the right side of the graphic.

AudioTest20130925.07_raw_tx



In this test, the caller spoke the word "Three" repeatedly, louder each time. The IVR recognized the word the on the indicated attempt. At this point, the caller was shouting.

AudioTest20130925.06_raw_tx



Additional tests on 9/27 were conducted using an Apple iPhone 4 over the AT&T network. DTMF tones peaked at -7 dB.

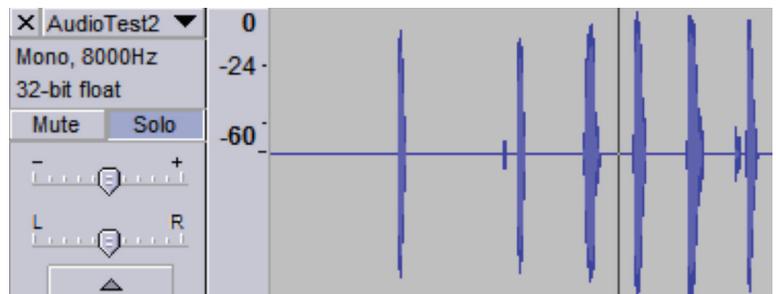
In this sample from the cell phone, the first three voice commands sounded distorted with digital artifacts. The fourth attempt sounded clear, and peaked at -20 dB. The next command (indicated by the arrow) was considerably louder and elicited a response from the IVR.

AudioTest20130927.04_raw_rx



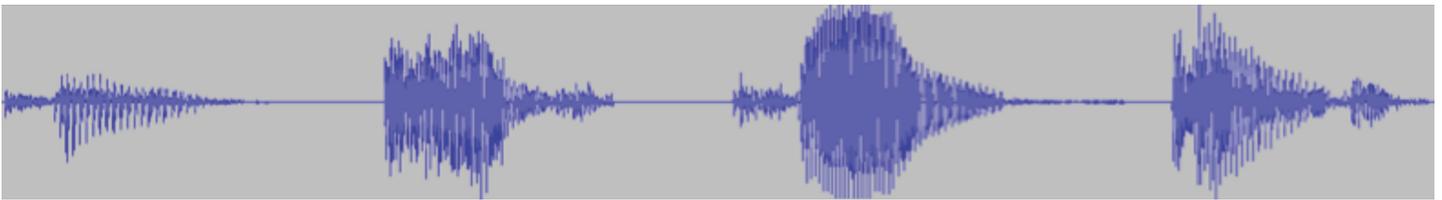
In this sample, audio levels varied up to a maximum of -5 dB. The IVR did not respond to any attempt. No audio distortion was noted.

AudioTest20130927.05_raw_rx



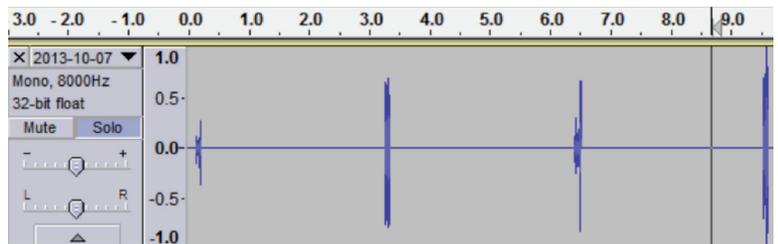
Comparison between the raw audio, and processed audio, show no significant change in volume level as a result of signal processing within the VoIP switch.

Testing on October 16 focused on analog telephone lines connected to a VoIP switch.



This capture from the analog 10/16/2013 shows the word "two" spoken four times at increasing volume levels. In the first and third samples, the "T" consonant sound can be clearly seen at the left edge of the wave form. The second and fourth repetitions show that the consonant sound has been clipped, resulting in an "ooh" sound . The IVR consistently fails to respond when the beginning of a word is clipped. This clipping action is taking place before the audio enters the switch.

A similar test on another analog line revealed both the leading and trailing edges of the wave forms clipped, resulting in unintelligible audio.



This wave form is extremely low level, and completely unintelligible. Even when normalized to 0vu, the speech was still garbled.

AudioTest20131016.04_tx



Conclusion

PCS recommends that the client turn to the PRI provider and ask that the audio levels be amplified to the same level as observed by DTMF tones. 10 dB seems to be a reasonable figure.

Distorted and clipped audio will cause the IVR to fail regardless of audio level. The audio captures above show that distortion is introduced into the audio stream before the audio reaches the switch. This can only be resolved by the Telco. Audio distortion due to cellular compression algorithms may distort the audio to the point where voice is not recognized at any audio level. We did not experience a failure with DTMF. If a caller can be identified that is experiencing DTMF failure, it may be valuable to repeat this analysis using that specific caller.

Personnel

Testing on behalf of Sample Company was carried out by these roles of the PCS team:

- **Software Developer** - role includes the development of virtual telephone software, which simulates the actions and signaling of actual ShoreTel desk sets, used with expanding ShoreTel functionality otherwise not possible.
- **Senior VoIP Engineer and Developer** - role includes implementation of new ShoreTel installations and working with Telco and network providers.
- **Senior Developer** - role focuses on design and implementation of IVR systems, including interfacing them to databases and other third-party vendors.

About PCS

Palitto Consulting Services are experts in integrating ShoreTel telephone systems with external systems and devices. The Palitto Library contains over 100 proven solutions for maximizing ShoreTel installations. PCS routinely creates custom solutions for clients with exacting requirements.



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