Voice Clipping ASP

Problem

Voice switches have inherent call connect delays. In air traffic control switches these delays cause significant problems on “voice call” lines. A controller at one facility can push a button and immediately start talking to controllers at other facilities. If the receiving switch delays are more than 100-200 msecs part of the message is lost. This is often referred to as “audio clipping.”

In some switches the delays can be as much as 500 msecs. The delay depends on several factors, such as switch design and the amount of traffic that is being handled by the switch. The busier the switch the more likely the delay will increase.

This situation can result in the loss of a word or part of a word – making it necessary for the receiver to guess what was said or ask for the message to be repeated. In many instances this can cause significant problems. In an air traffic control situation this adds to the work load of a very busy controller and potentially decreases safety.

Solution

Digital Signal Products, Inc. has developed an Audio Signal Processor (ASP) that totally removes this problem. The processor is “transparent” to the switch and all other operations proceed normally. The following is a brief outline of how the ASP works:

It monitors the receive lines for voice activity - speech energy is compared to idle channel noise to detect a valid incoming call.

When the ASP detects an incoming call it immediately starts storing the incoming audio. Within 5-10 msec the ASP sends a Zip tone to the switch to announce an incoming voice call.

The ASP stores up to 500 msec of incoming audio. After 500 msec of initial delay, it starts sending this stored voice to the switch from the very beginning, thus giving sufficient time for the receiving switch to connect the incoming voice call to the designated position.

When an operator answers the incoming call, the ASP starts looking for a pause in the incoming voice, and removes the initial delay. Therefore, the ASP incorporates 500 msec of delay ONLY during the initial call connect period. This process is totally transparent to the user. There is no delay for outgoing calls.

In case of an ASP failure a there is an automatic by-pass that permits normal communication without any loss of service.

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Configuration

The ASP's are housed in a 19-inch rack mountable chassis, and occupies only 5.25 inches of rack space. The Voice Clipping ASP design employs a fully modular approach, using state of the art Digital Signal Processor (DSP). Each Audio Signal Processor (ASP) card contains a DSP and necessary circuitry to provide an interface for two voice call lines. To ensure there are no single point failures, each ASP card works in an independent manner to enhance overall system reliability. Each ASP provides a red LED and a failure alarm contact closure output. A total of twelve cards can be installed in a single chassis to remove voice clipping on 24 voice call lines. Interconnect to the ASP chassis is done through a backplane that provides 25 pair Amp-Champ telephone connectors, using industry standard, readily available mass terminated telephone cables.

Specifications

Input/Output Impedance: 600 Ohms, Balanced, transformer coupled to reduce hum and noise
Input Level: -30 to +7 dBm
Overall Gain: +/- 2 dB
Frequency Response: 300-3000 Hz +/- 2 dB with-respect-to 1004 Hz
Adjustments: None required
Distortion: 2% maximum
Coding: PCM, mu-law, 64 Kbits/s
Initial Delay: 400/500 msec, switch selectable; voice call delays up to 1000 msec are available
VOX Threshold: -40 dBm maximum
Cross Talk between channels: -65 dB minimum
Idle Channel Noise Output: 20 dBrnCO
Front panel LEDs for voice activity, line busy and health status
4-Wire Echo Return Loss: 20 dB minimum @ 600 Ohms

Other Related Products

Best Signal Selection (BSS) – In some radio communication systems, there are more than one receiver for a given frequency. This occurs for reasons such as too great a distance or terrain interference with only one receiver. Under some circumstances, multiple receivers will have a signal that exceeds an audible threshold, but the strength and quality may be significantly different. The Best Signal Selection ASP monitors all receivers and within 70-250 msecs routes the call using the highest quality signal.

Tone Notching and Automatic Gain Control (AGC) – Air Traffic Controllers have identified two significant problems in many current voice switches: high frequency tones accidentally induced into their headsets, and improper balancing of volume of incoming calls. DSP has developed an Audio Signal Processor (ASP) that can remove two extraneous tones simultaneously in less than 70 msec without degrading the quality of the incoming signal. The same ASP also adjusts all incoming calls to a volume that is selected by the air traffic controller.

Radio Control Equipment (RCE) – DSP has developed an RCE that provides full duplex voice and data communications between a control site and a remote site over unconditioned telephone lines and satellite links. The control end modem trains the remote end modem to achieve the highest possible data transmission rate. The operator voice bandwidth is 300-3400 Hz end-to-end. The design employs speech encoding at 8 Kbits/s.

Digital Signal Products, Inc.

Digital Signal Products, Inc. provides both high quality products for communication systems as well as services for developing solutions for specific customer problems. DSP can help establish requirements, provide both a high level design and a detailed design that satisfy a set of requirements and, if desired, actually build and test products that conform to the design.

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