ZLS38233

Advanced 4 Channel Voice Echo Canceller with DTMF Transceiver (Tx/Rx)

Firmware Data Sheet

April 2009

Features

- Independent 4 channel echo cancellation with programmable echo tail length (maximum combined echo tail length of 256 ms)
- Fully compliant to ITU-T G.165, G.168 (2000) and (2002) specifications
- PCM coding, μ/A-Law ITU-T G.711 or sign magnitude
- Per channel Fax/Modem G.164 2100 Hz and G.165 2100 Hz phase reversal Tone Disable
- Per channel echo canceller parameters control
- Transparent data transfer and mute
- Fast reconvergence on echo path changes
- Fully programmable convergence speeds
- Patented Advanced Non-Linear Processor with high quality subjective performance
- Protection against narrow band signal divergence and instability in high echo environments
- Offset nulling of all PCM channels
- -24 to +21 dB - 3 dB step level adjusters at all signal ports
- G.169 ALC (automatic level control) maintains loop gain through a programmable gain range from –24 to 24 db
- Four independent DTMF Receivers (DTMF Rx)
- Four independent programmable tone/DTMF generators (DTMF Tx)

Applications

- VoIP gateways
- Wireless GSM/CDMA Media Gateways
- Wireless base stations
- Wireless and Wireline PBX
- DCME, Satellite multiplexer systems

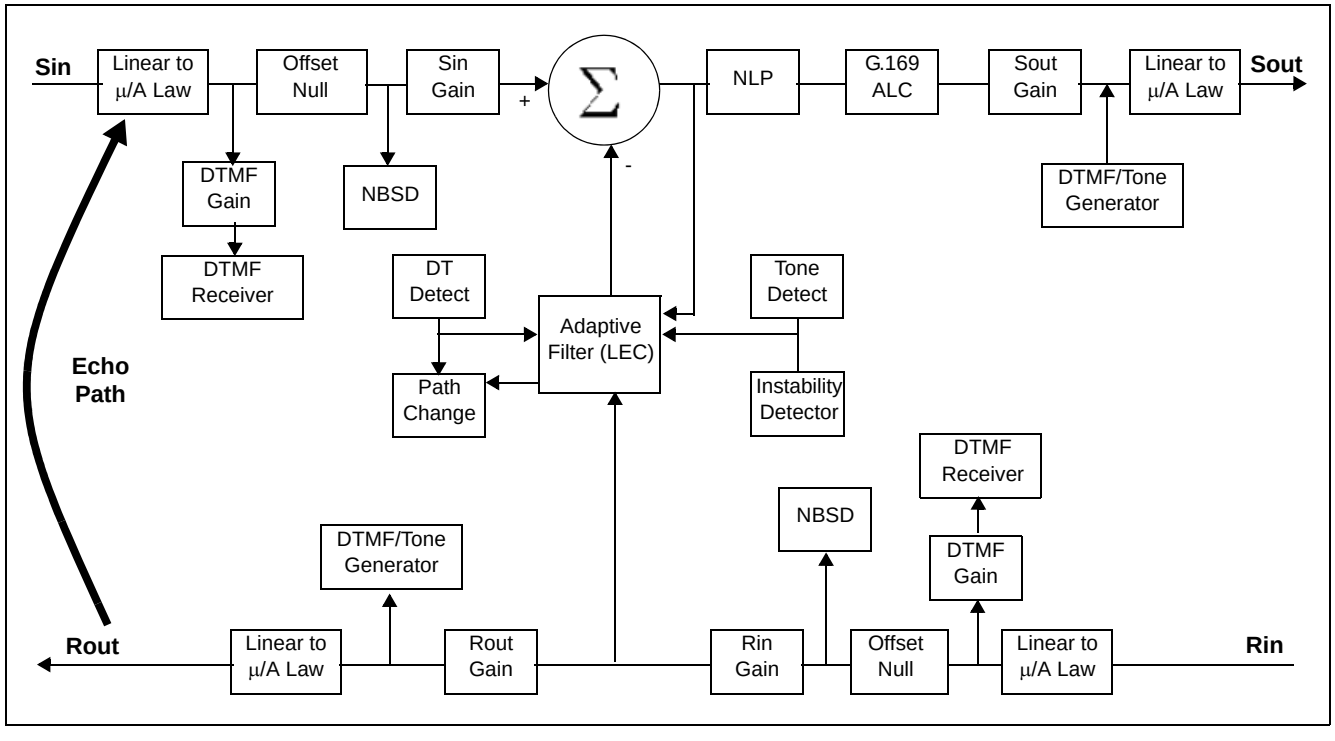

Figure 1 - Single Channel of ZL38233 VEC and DTMF Transceiver

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1.0 Description

The ZLS38233 Voice Echo Canceller firmware implements a cost effective solution for telephony voice-band echo cancellation conforming to ITU-T G.168 requirements. The ZLS38233 architecture contains 4 echo cancellers and 4 DTMF Tone Generators and Receivers. Each channel echo tail length is programmable to a maximum of 64 ms. The ZLS38233 also supports ITU-T G.165 and G.164 tone disable requirements.

Each echo canceller contains the following main elements (see Figure 1).

- Adaptive Filter for estimating the echo channel with programmable filter length)
- Subtractor for cancelling the echo
- Double-Talk detector for disabling the filter adaptation during periods of double-talk
- Path Change detector for fast reconvergence on major echo path changes
- Instability Detector to combat instability in very low ERL environments
- Patented Advanced Non-Linear Processor for suppression of residual echo, with comfort noise injection
- DTMF (RX) receiver
- DTMF TX /Programmable tone generator
- Disable Tone Detectors for detecting valid disable tones at send and receive path inputs
- Narrow-Band Detector for preventing Adaptive Filter divergence from narrow-band signals
- Offset Null filters for removing the DC component in PCM channels
- -24 to +21 dB - 3 dB step level adjusters at all signal ports
- Parallel controller interface compatible with Motorola microcontrollers
- PCM encoder/decoder compatible with μ/A-Law ITU-T G.711 or Sign-Magnitude coding
- Each echo canceller in the ZLS38233 has four functional states: *Mute*, *Bypass*, *Disable Adaptation* and *Enable Adaptation*. These are explained in the section entitled Echo Canceller Functional States.

2.0 Voice Echo Cancellation

The following section describes the operation each block of the voice echo canceller.

2.1 Adaptive Filter

The adaptive filter adapts to the echo path and generates an estimate of the echo signal. This echo estimate is then subtracted from Sin. The adaptive filter is a 2048 tap FIR adaptive filter which is shared by the 4 echo cancellers. Each section can be program the FIR filter length from 128 (16 ms) to up to 512 taps (64 ms) but the total for all 4 echo cancellers can not exceed 2048 taps (256 ms).

2.2 G.168 Voice Echo Canceller (VEC) Description

The VEC filter adapts to the echo path and generates an estimate of the echo signal. This echo estimate is then subtracted from the echo path. The total echo tail length of the VEC is programmable up to 256 ms in 8 ms steps (Note: this 256 ms maximum capacity is shared among the four channels).

Figure 2 shows a possible line echo profile. There are three reflections: one at the closest central office, the second at the far end central office and the third at the far end telephone. The VEC algorithm is a full band echo canceller, which means that it will cancel all echo within the programmed echo tail length (i.e., less than 60 ms in Figure 2). This is superior to other line echo cancellers that commonly use a sliding window function and are not able to cancel all echo's within the 64 ms echo profile as shown.

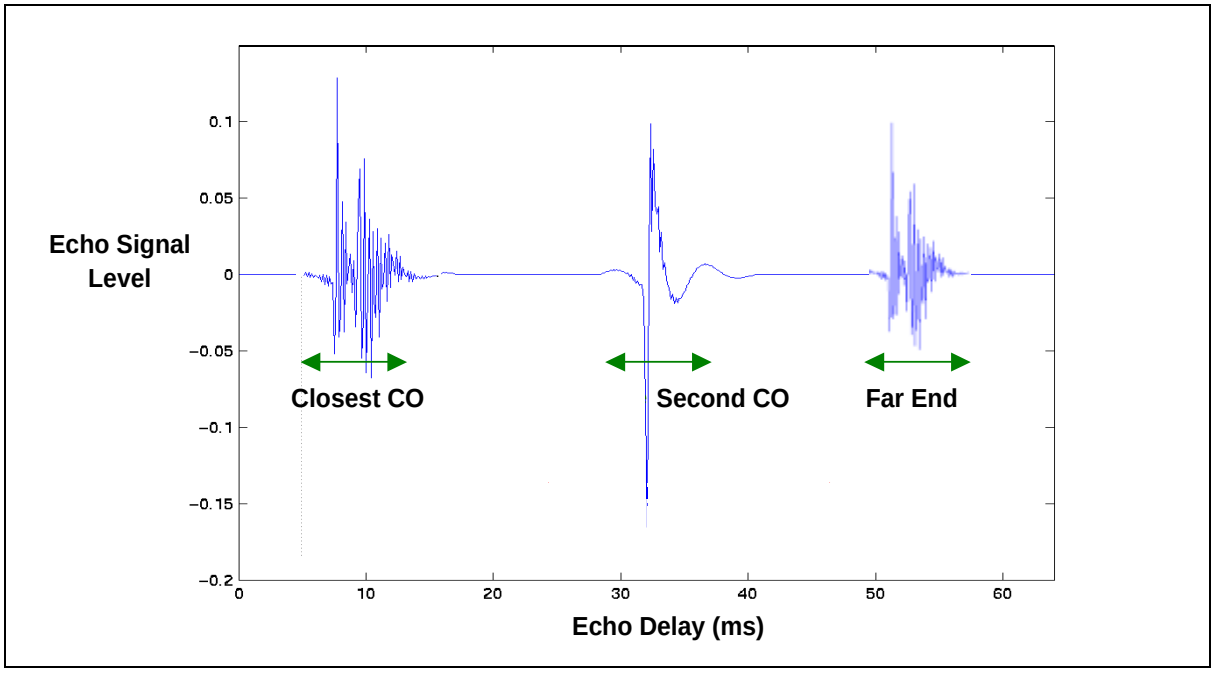

Figure 2 - Typical Line Echo Profile

2.3 Double-Talk Detector

Double-Talk is defined as those periods of time when signal energy is present in both directions simultaneously. When this happens, it is necessary to disable the filter adaptation to prevent divergence of the Adaptive Filter coefficients. Note that when double-talk is detected, the adaptation process is halted but the echo canceller continues to cancel echo using the previous converged echo profile. A double-talk condition exists whenever the relative signal levels of Rin (Lrin) and Sin (Lsin) meet the following condition:

$$Lsin > Lrin + 20log_{10}(\text{VEC\_ERL\_EST})$$

In the G.168 standard, the echo return loss is expected to be at least 6 dB. This implies that the Double-Talk Detector Threshold should be set to 0.5 (-6 dB). However, in order to achieve additional guardband, the Double-Talk Detector Threshold is set internally to 0.5625 (-5 dB).

In some applications the return loss can be higher or lower than 6 dB. The ZLS38233 allows the user to change the detection threshold to suit each application's need.

2.4 Path Change Detector

Integrated into the ZLS38233 is a Path Change Detector. This permits fast reconvergence when a major change occurs in the echo channel. Subtle changes in the echo channel are also tracked automatically once convergence is achieved, but at a slower speed.

2.5 Non-Linear Processor (NLP)

When the VEC functions have converged and a single talk (i.e., Rin to Rout incident voice signal; no return Sin to Sout voice, or Sin to Sout incident voice signal; no return Rin to Rout voice) condition exists there will be low level echo on the return path due to quantization errors. The VEC echo cancellation algorithms cannot predict quantization errors; therefore, this small amount of echo (i.e., residual echo) will not be cancelled. The NLP algorithm, illustrated in Figure 3, replaces this residual echo signal with a background noise signal that diminishes to zero with time. This background noise has the same signal strength and same spectral characteristics as the original signal.

During normal operation when the input signal level of the NLP is below the background noise level, the filter of Figure 3 will learn the background noise level through switch SW1. When the input signal level is above the background noise level and the signal is residual echo, the Filter input will be zero through SW1 and the filter output signal will pass through SW2 to Sout. When the input signal level is above the background noise level and the signal is no dominated by residual echo SW2 will bypass the NLP and the filter input will be connected to zero.

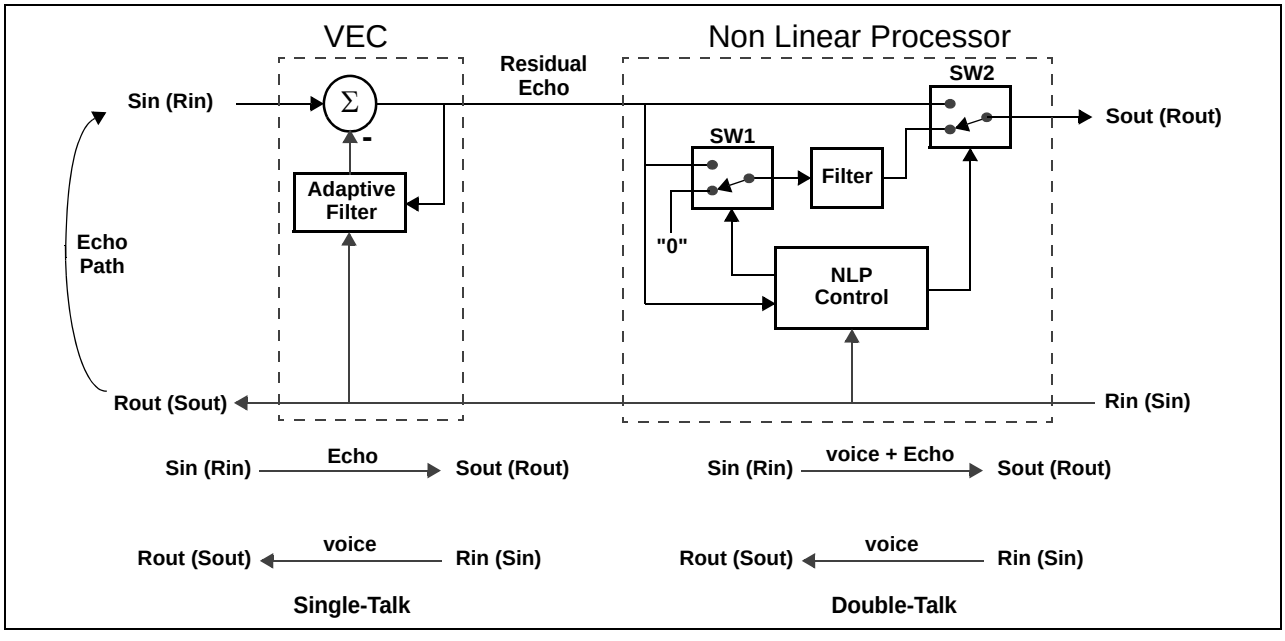

Figure 3 - NLP Block Diagram

2.6 Instability Detector

In systems with very low echo channel return loss (ERL), there may be enough feedback in the loop to cause stability problems in the adaptive filter. This instability can result in variable pitched ringing or oscillation. Should this ringing occur, the Instability Detector will activate and suppress the oscillations.

2.7 Narrow Band Signal Detector (NBSD)

Single or dual frequency tones (i.e. DTMF tones) present in the receive input (Rin) of the echo canceller for a prolonged period of time may cause the Adaptive Filter to diverge. The Narrow Band Signal Detector (NBSD) is designed to prevent this by detecting single or dual tones of arbitrary frequency, phase, and amplitude. When narrow band signals are detected, adaptation is halted but the echo canceller continues to cancel echo.

2.8 Offset Null Filter

Adaptive filters in general do not operate properly when a DC offset is present at any input. To remove the DC component, the ZLS38233 incorporates Offset Null filters in both Rin and Sin inputs.

2.9 Adjustable Level Pads

The ZLS38233 provides adjustable level pads at Rin, Rout, Sin and Sout. This setup allows signal strength to be adjusted both inside and outside the echo path. Each signal level may be independently scaled with anywhere from -24 to +21 dB in 3 dB steps. Level values are set using the Gains register.

CAUTION: Gain adjustment can help interface the ZLS38233 to a particular system in order to provide optimum echo cancellation, but it can also degrade performance if not done carefully. Excessive loss may cause low signal levels and slow convergence. Exercise great care when adjusting these values. Also, due to internal signal routing it is not recommended that gain adjustments be used on Rin or Sout in this mode.

2.10 Echo Canceller Functional States

Each echo canceller has four functional states: **Mute, Bypass, Disable Adaptation** and **Enable Adaptation**.

2.10.1 Mute

In Normal configuration writing muting the Rin path will replaces Rin with quiet code which is applied to both the Adaptive Filter and Rout. Writing muting the Sin path replaces the Sout PCM data with quiet code.

	LINEAR 16 bits 2's complement	**SIGN/ MAGNITUDE μ-Law A-Law**	**CCITT (G.711)**	
			μ-Law	**A-Law**
+Zero (quiet code)	0000_{hex}	80_{hex}	FF_{hex}	$D5_{hex}$

Table 1 - Quiet PCM Code Assignment

2.10.2 PCM and Voice Bypass

The Bypass state directly transfers PCM codes from Rin to Rout and from Sin to Sout. **When Bypass state is selected, the Adaptive Filter coefficients are reset to zero.** Bypass state must be selected for at least one frame (125 ms) in order to properly clear the filter.

2.10.3 Disable Adaptation

When the Disable Adaptation state is selected, the Adaptive Filter coefficients are frozen at their current value. The adaptation process is halted, however, the echo canceller continues to cancel echo.

2.10.4 Enable Adaptation

In Enable Adaptation state, the Adaptive Filter coefficients are continually updated. This allows the echo canceller to model the echo return path characteristics in order to cancel echo. This is the normal operating state.

2.11 ZLS38233 Throughput Delay

The throughput delay of the ZLS38233 varies according to the device configuration. For all device configurations, Rin to Rout has a delay of two frames and Sin to Sout has a delay of three frames. In Bypass state, the Rin to Rout and Sin to Sout paths have a delay of two frames.

2.12 ITU-T G.168 Compliance

The ZLS38233 has been certified G.168 (1997), (2000) and (2002) compliant in all 64 ms cancellation modes by in-house testing with the GL comm echo canceller tester.

3.0 Additional Features

The following section describe the addition features incorporated in the ZLS38233. These include:

- G.169 Automatic Level Control
- Disable Tone Detector
- Tone Generator (DTMF TX)
- DTMF Detection (RX)

3.1 G.169 Automatic Level Control

This block automatically controls the level of the Sin to Sout path located at the output of the echo canceller. The user is given the option for gain control ranges. The the gain range adjustment range from ‘-24 dB’ to ‘+24 dB’.

3.2 Disable Tone Detector

The G.165 recommendation defines the disable tone as having the following characteristics: 2100 Hz (±21 Hz) sine wave, a power level between -6 to -31 dBm0, and a phase reversal of 180 degrees (± 25 degrees) every 450 ms (±25 ms). If the disable tone is present for a minimum of one second with at least one phase reversal, the Tone Detector will trigger.

The G.164 recommendation defines the disable tone as a 2100 Hz (±21 Hz) sine wave with a power level between 0 to -31 dBm0. If the disable tone is present for a minimum of 400 ms, with or without phase reversal, the Tone Detector will trigger.

The ZLS38233 has two Tone Detectors per channels (for a total of 8) in order to monitor the occurrence of a valid disable tone on both Rin and Sin. Upon detection of a disable tone, TD bit of the Status Register will indicate logic high and an interrupt will be generated.

There are two disable tone detection controls associated with the disable tone detector. The primary controls the selection between G.165 and G.164 tone disable.

In response to a valid disable tone, the echo canceller must be switched from the Enable Adaptation state to the Bypass state. This can be done in two ways, automatically or externally. In automatic mode, the Tone Detectors internally control the switching between Enable Adaptation and Bypass states. This can be done externally or automatically. In external mode, an external controller is needed to service the interrupts. Following the detection of a disable tone on a given channel, the external controller must switch the echo canceller from Enable Adaptation to Bypass state.

The secondary controls are used in applications where both G.164 and G.165 disable tones need to be detected. Under these conditions the primary control must be set to G.165. All the primary interrupt control work as described. The secondary control allow an interrupt to be generated if a G.164 tone is received.

Once a Tone Detector has been triggered, there is no longer a need for a valid disable tone (G.164 or G.165) to maintain Tone Detector status (i.e., TD bit high). The Tone Detector status will only release (i.e., TD bit low) if the signals Rin and Sin fall below -30 dBm0, in the frequency range of 390 Hz to 700 Hz, and below -34 dBm0, in the frequency range of 700 Hz to 3400 Hz, for at least 400 ms. Whenever a Tone Detector releases, an interrupt is generated (i.e., $\overline{IRQ}$ pin low).

3.3 Tone Generator (DTMF TX)

The ZLS38233 firmware incorporates a programmable tone generator. The following features are supported in the tone generator:

- Ability to generate dual or single tone with pre-programmed frequencies.
- Ability to generate ringer tone. The warble effect is generated using a square wave with programmable duty cycle and frequency.
- Two sets of eight pre-programmed frequencies are provided in user config file where each pair of frequencies can be selected by tone generator register. Overall 16 frequencies can be selected by tone generator. All frequencies are programmable by users. The default values are those frequencies used in DTMF, ring and dial tone etc.
- Burst mode support with programmable burst time (from approximately 0 to 4 seconds).
- Programmable output gain from -18 to 24 dB with 6 dB steps. (Default is 0 dB)
- Individual output gain for low and high tones.
- DTMF output can be routed to Rout or Sout (or both) outputs. Depending on the selection, during DTMF injection, actual signal passing through Sout or Rout ports will be muted.
- -2.1 dB twist support mode for DTMF tones. DTMF algorithm is based on generating two sinusoidal tones with frequencies specified in Table 2.

	1209 Hz	**1336 Hz**	**1477 Hz**	**1633 Hz**
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

Table 2 - DTMF Generation Standard Frequency Table

An optional -2.1 dB twist will be applied to lower tones when the TWENB bit is set. For tone ringer same DTMF function can be used with the difference that output will toggle between two single tones with a programmable frequency and a known warble rate.

3.4 DTMF Detection (RX)

This block detects the 16-DTMF digits DTMF digits (0-9, #,*, A,B,C, D) and talk-off abatement compliant to Q.24, Bellcore, and Mitel tape tests. The ZLS38233 will issue a GPIO-based interrupt to the host to indicate a successful DTMF detection. The minimum “must-detect” signal duration is 40 msecs, while the minimum pause duration is 30 msecs. Therefore, the host should check for DTMF detection within 70 msecs upon receiving a successful DTMF detection interrupt.

3.4.1 DTMF Detector Performance Measurements

The table below describes the various DTMF RX specifications:

Test Type	Performance	Performance (NTT)
Frequency Deviation	+/-1.5% accept; +/-3.5% reject	+/-1.8% accept; +/-3.0% reject
Minimum Tone Duration	40 ms accept; 23 ms reject	30 ms accept; 24 ms reject
Minimum Interdigital Interval (Pause Duration)	40 ms	30 ms
Accept Levels (DTMF Sensitivity)	> -24 dBm accept; < -55 dBm reject	> -25 dBm accept; < -29 dBm reject
Twist (ratio of high group-power to low)	-8 to +4 dB	-8 to +4 dB
SNR (accept digits above)	14 dB	14 dB

Table 3 - DTMF RX Specifications

Each DTMF receiver has an input gain pad. This pad is used to set the input signal level appropriately for different DTMF requirements. The default setting (gain pad se to 0 dB) will meet the AT&T requirements which has a minimal detection level of -25 dBm and a must reject signal level of -55 dBm. For NTT these levels are -24 dBm and -29 dBm respectively. Programming a value of -5 dB into the gain pads will allow the DTMF receiver to pass NTT.

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