

Features

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- Selectable Acoustic and Line Echo Cancellers (AEC & LEC)
- Programmable echo tail cancellation length from 8 to 256 ms
- Noise Reduction - up to 20 dB for white noise and up to 30 dB in a typical car environment
- Non-linear Echo Controller
- 16 band equalizer in the Sout and Rout path
- Handles up to 6 dB line echo return loss measured from Sout to Rin
- No divergence due to double-talk (patented Adaption Control)
- Programmable gain at Rin, Sin and Sout to accommodate system requirements
- User gain control provided on speaker range (Rout) from -21 dB to +24 dB in 3 dB steps
- Programmable DC offset nulling filter cutoff frequency on send and receive paths
- 16-bit linear coding at acoustic interface (Rout & Sin)
- Programmable coding (μ /A-law ITU-T G.711 or 8-bit sign magnitude or 16-bit linear) at Line interface (Sout & Rin)
- Superior voice quality with patented Non-Linear Processor (NLP)
- Patented Clipping Compensation
- Patented Narrow-band signal divergence protection
- Provides mute capability on Rout and Sout independently
- Patented Anti-howling - prevents oscillation in echo canceller audio path
- Speaker path AGC (Rout)
- G.169 ALC in the Sin to Sout path
- Application specific tuneable performance
- Run-time programmable parameters

Applications

- Handsfree car kits

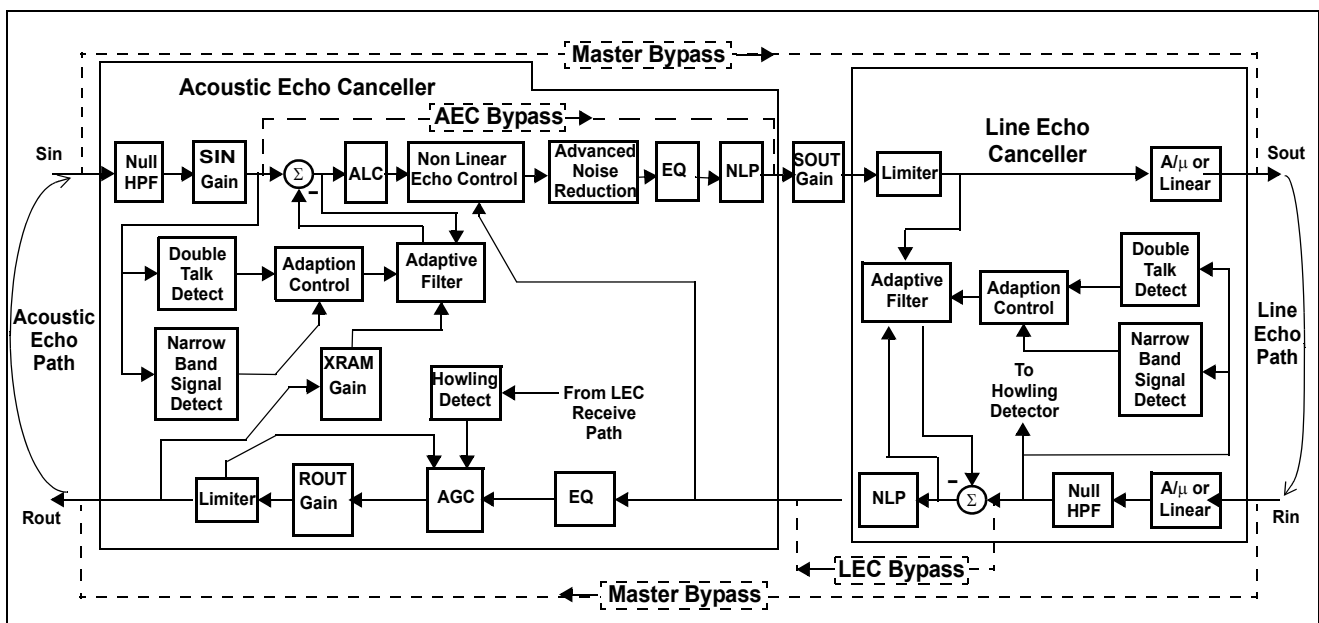


Figure 1 - Firmware Block Diagram

1.0 Functional Description

ZLS38500 is firmware that runs on the ZL38004 (Advanced Acoustic Echo Canceller with Noise Reduction) hardware platform to realize audio applications requiring acoustic echo cancellation and noise reduction.

2.0 Firmware Algorithm Description

The ZLS38500 application firmware contains two echo cancellers which can be configured to provide one channel of echo cancellation in back to back mode. One for echo canceller to cancel Acoustic echo and the other to cancel line echo, providing full-duplex echo cancellation.

The echo canceller used on acoustic path is called acoustic echo canceller (AEC) and the one used on line echo path is called line echo canceller (LEC).

Each block can be bypassed allowing a AEC or LEC mode only. The intended platform (ZL38004) has a 256 ms filter that can be divided up between the AEC and the LEC. If the firmware is configured in AEC only mode then the AEC can cancel echo tails up to 256 ms. If the firmware is configured with both AEC and LEC active (in a back to back configuration) then the 256 ms must be shared between the AEC and LEC (i.e., 192 ms for AEC and 64 ms for LEC). The echo tail lengths are programmable in 8ms blocks.

The ZLS38500 firmware implements the following functions:

- **Adaptive Filter:** continuously adapts to the echo path and uses this data to cancel the acoustic or line side echo
- **NLP:** Non-Linear Processor reduces residual echo to a negligible level
- **Double-Talk Detector:** detects double-talk conditions so they do not adversely affect the NLP and adaptive filter operations
- **Non-Linear Echo Control:** is provided to compensate for non linearities in the speaker/mic circuitry
- **Advanced Noise Reduction:** reduces background noise from the near-end speech signal (Sin)
- **Rout Equalizer:** a 16 band equalizer is placed in the Rout path to compensate for bandwidth limitations in the speaker
- **Sout Equalizer:** a 16 band equalizer is placed in the Sout path
- **G.169 ALC:** Automatic Level Controller in the Sin to Sout path
- **AGC:** Automatic Gain Control for the speaker output (Rout).
- **Gain Pads:** used to adjust signal levels at Sin, Sout and the AEC adaptive filter. The Sin Gain Pad is adjustable from -24 to +21 dB in 1.5 dB steps; Gain Pads for Sout and XRam are adjustable from -24 to +21 dB in 3 dB steps.
- **User Gain (Rout Gain):** adjusts speaker volume at Rout from -24 to +21 dB in 3 dB steps
- **Clipping Compensation:** adjusting the analog and digital gain stages to minimize clipping at the audio input
- **Narrowband Detector:** detects narrowband signals coming from the acoustic or line side interfaces. Prevents narrowband signals from adversely affecting adaptive filter operation
- **Offset Nulling filter:** removes DC signals from the Rin and Sin inputs
- **Limiters:** used to limit Rout and Sout signals to defined maximum values
- **Howling detector:** detects howling caused by positive feedback in the loop. When howling is detected, the AGC gain is reduced to eliminate the howling.
- **A/ μ -Law or Linear:** converts 8-bit μ -law, A-law or sign magnitude PCM to 16-bit linear data, or passes 16-bit linear data unaltered. The A/ μ -Law converters are only on the Rin and Sout ports.

2.1 Acoustic Echo Canceller (AEC) Description

The Acoustic echo canceller section contains two functional blocks, a linear acoustic echo canceler and a non linear echo controller. The linear echo canceller is used to remove the linear portion of the echo produced by the direct coupling between the speaker and microphone and the room acoustics (this is majority of the acoustic echo). The non-linear echo controller is used to remove non linear echo caused by the speaker (produced when the speaker is driven to maximum capability).

2.2 Line Echo Canceller (LEC) Description

The LEC 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 echo tail length of the LEC is programmable up to 256 ms in 8 ms steps (Note: this 256 ms maximum capacity is shared with the AEC).

2.3 Adaptive Filter

The adaptive filter adapts to the echo path and generates estimate of the echo path. This replica is subtracted from the return signal. The resulting echo cancellation is monitored and the estimated echo updated.

2.4 Non-Linear Processor (NLP)

When the AEC/LEC functions have converged and a single talk (i.e., Rin to Rout incident audio signal; no return Sin to Sout audio, or Sin to Sout incident audio signal; no return Rin to Rout audio) condition exists there will be low level echo on the return path due to ADC and DAC quantization errors. The AEC/LEC 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 2, replaces this residual echo signal with a comfort noise signal that 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 2 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 not dominated by residual echo SW2 will bypass the NLP and the filter input will be connected to zero.

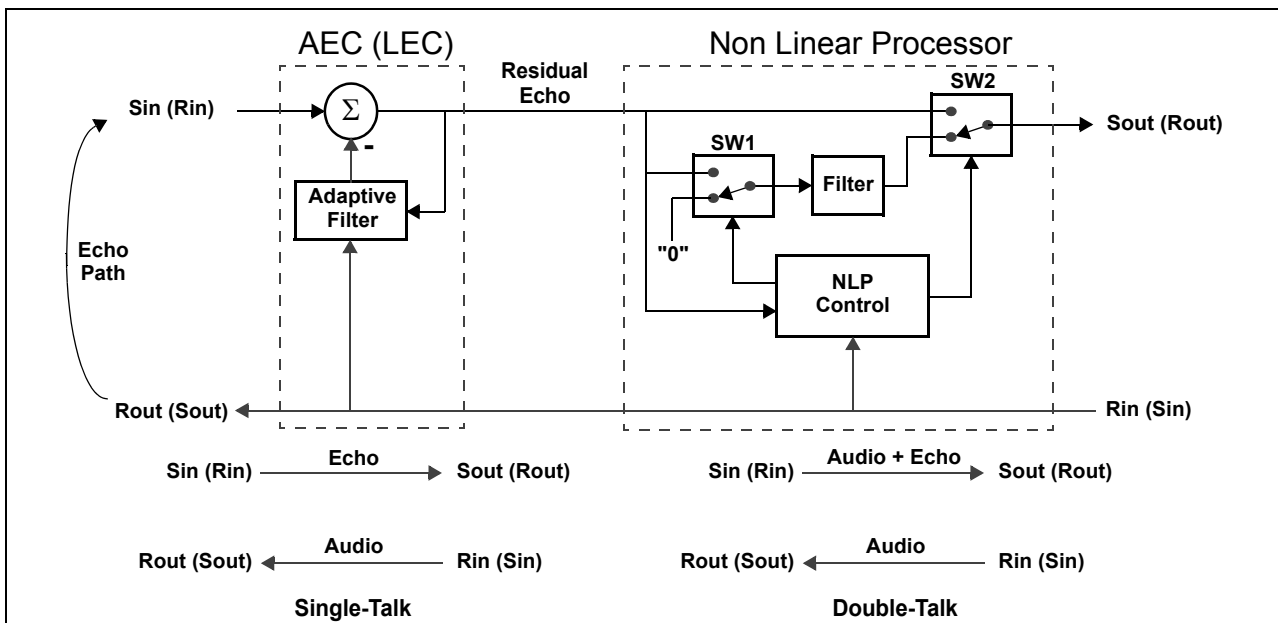


Figure 2 - NLP Block Diagram

2.5 Noise Reduction

The performance of a hands free car kit depends heavily on the ambient or background noise level in the passenger compartment of a moving car. Wind, engine and road noise, which are the main sources of this ambient noise, can severely deteriorate the voice quality of near-end talker speech. Road noise is caused by tire tread impacting the surface of the road; wind noise is caused by air hitting the surfaces of the car and flowing over irregular shapes like rear view mirrors; and engine noise is caused by the periodic combustion of fuel within the engine cylinders. Therefore, an ambient noise reduction function is required to remove this noise from the near-end speech signal.

The ZLS38500 Noise Reduction block is composed of two elements, a Clipping Compensation block and Spectral Noise Reduction block.

2.6 Clipping Compensation

Wind noise is caused by air flowing across the surfaces of a moving vehicle and by the air flow due to large pressure differences between the inside and outside of a moving car body. The ZLS38500 includes a Noise Reduction algorithm to mitigate the effect of wind noise, as well as road noise, on speech sound quality, see Figure 3. However, high sound pressure wind noise (e.g., car speed of 100 km/h (60 mph) with windows open) can cause the analog interface hardware of the ZLS38500 (i.e., Mic Amplifier & CODEC) to be saturated. When this happens analog voice signals converted by the CODEC ADC will be distorted or clipped. This will inhibit both the Acoustic Echo cancellation and Noise Reduction circuits from performing properly.

A "Smart AGC" has been implemented, through ZLS38500 Firmware, on the microphone input (Sin) to eliminate this clipping condition while maintaining the gain settings selected by the user. This is achieved by simultaneously adjusting the analog and digital gain stages (Figure 3) to minimize clipping at the audio input. This method maintains the maximum SNR and microphone sensitivity for a particular environment, and is superior to implementing a low fixed microphone gain followed by a gain pad after the noise reduction section. This algorithm also compensates for abrupt echo path changes (Smart AGC gain adjustment) that would normally initiate reconvergence.

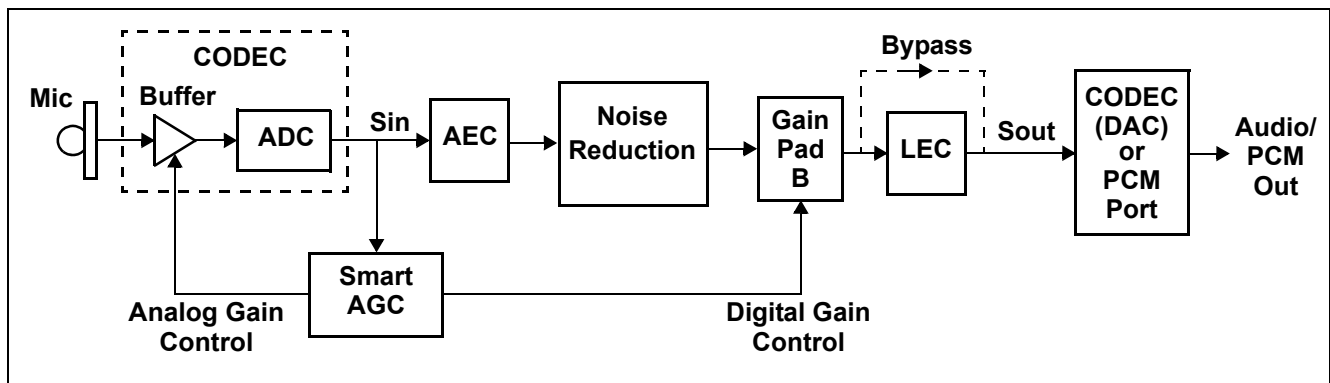


Figure 3 - Clipping Compensation Circuit

2.7 Spectral Noise Reduction

Figure 4 shows the block diagram of the noise reduction algorithm. The noisy speech signal is first converted to the frequency domain and then signal power levels are calculated for individual frequency bins. The Speech signal power and Noise signal power are estimated based on the power level calculation for each bin. These two new estimated power levels are used to adaptively generate noise reduction coefficients that are applied to the noisy speech signal. This result is converted back to the time domain to produce a Noise Reduced Output.

The noise reduction level is programmable so the user can adjust the level of noise cancellation to meet system requirements (default =13 dB, based on white noise). A larger value in this register will increase the amount of

noise reduction. As the amount of noise reduction is increased, the amount of distortion in the audio path also increases. Noise reduction can be increased to 20 dB (based on white noise) without significant distortion of the voice signal. In the passenger compartment of a car the noise source is concentrated in the lower portion of the frequency spectrum (typically below 250 Hz). This allows the noise reduction to be increased to approximately 30 dB.

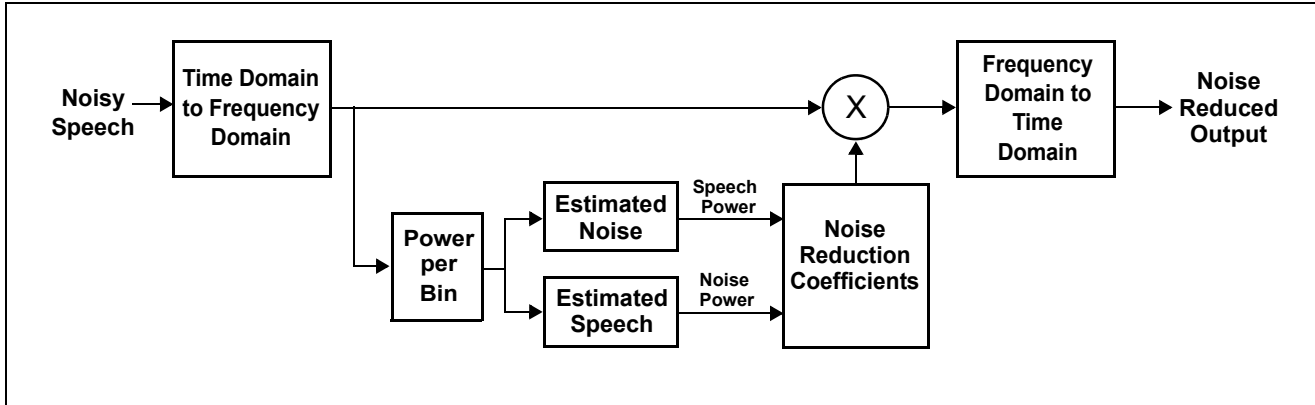


Figure 4 - Noise Reduction Block Diagram

2.8 Narrow Band Signal Detector (NBSD)

Narrow-band signals such as single or multi-frequency tones (i.e., DTMF or signalling tones) present at the reference input of an 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 divergence by first detecting single or multi-frequency tones of arbitrary frequency, phase, and amplitude. When these narrow-band signals are detected, the echo canceller continues to cancel echo, but the filter adaptation process is stopped.

2.9 Howling Detector (HWLD)

The Howling detector is part of an Anti-Howling control algorithm that prevents oscillation as a result of positive feedback on the audio paths. When howling is initially detected the AGC at Rout will reduce the signal level by 6 dB. If howling persists then the AGC will decrease the signal level by an additional 6 dB. This continues until the howling stops. At this point the AGC will increase the gain to determine if the howling begins again. If it does then the AGC gain is reduced again. If the howling condition does not exist then the AGC will increase its gain in 6 dB steps until the user defined signal level (without howling) is reached.

2.10 Programmable High Pass Filter (Offset Nulling Filter)

Programmable high pass filters are placed at the Sin and Rin ports, see figure 1. These filters have two functions: to remove any DC offset that may be present on either the Rin or the Sin ports; and second, to filter low frequency (below 100 Hz) noise such as road noise.

2.11 Limiters

The limiters on the send and receive paths (See Figure 1) prevent the Rout and Sout output signal levels from exceeding a preset limit. This is to ensure that the linear region of the speaker is not exceeded. Limiter values are programmable. See registers Upper limit value for Sout and Upper limit value for Rout.

2.12 User Gain

The User Gain function is used to adjust the speaker volume outside the echo path. This gain is optimally placed outside the echo path so re-convergence is not necessary after gain changes. It is important to use **ONLY** this gain pad to adjust the speaker volume during normal operation. This gain is adjustable from -24 to +21 dB in 3 dB steps.

2.13 Automatic Gain Control (AGC)

The Automatic Gain Control (AGC) is used to automatically adjust the input speech level of the audio signal to a predetermined output value. The AGC automatically decreases the speaker path gain during the following conditions:

- when clipping of the receive signal occurs at Rout,
- when initial convergence of the acoustic echo canceller detects unusually large echo return
- when howling is detected

2.14 Programmable Gain Pad

The AEC has gain pads associated with Sin, Sout and the AEC adaptive filter (Xram). These gain pads are intended to be set once during initialization and not adjusted during normal operation. Their purpose is to help fine tune the performance of the acoustic echo canceller for a particular application. Sin gain pad is adjustable from -24 to +21 dB in 1.5 dB steps; Sout and Xram gain pads are adjustable from -24 to +21 dB in 3 dB steps.

2.15 Mute Function

Mute functions are provided for independent control of the Receive (Rin to Rout) and Send (Sin to Sout) audio paths. Setting the MUTE_R or MUTE_S bits causes quiet code to be transmitted on the Rout or Sout ports respectively.

LINEAR 16 bits 2's complement	SIGN/ MAGNITUDE μ /A-Law	CCITT (G.711)	
		μ -Law	A-Law
0x0000	0x80	0xFF	0xD5

Table 1 - Quiet PCM Code Assignment

2.16 Master Bypass

When the Master Bypass function is active, PCM data passes transparently from Rin to Rout and from Sin to Sout, with bit-wise integrity preserved. When this mode of operation is selected, most internal functions are automatically powered down to reduce power consumption.

2.17 AEC Bypass

The AEC bypass function allows send and receive PCM data to pass around the AEC, non linear echo controller, NLP, ALC and noise reduction function) with bit-wise integrity preserved (i.e., the echo estimate from the AEC adaptive filter is not subtracted from the Send path). This mode of operation does not affect the normal operation of other functions.

2.18 LEC Bypass

The LEC bypass function allows send and receive PCM data to pass around the LEC function with bit-wise integrity preserved (i.e., the echo estimate from the LEC adaptive filter is not subtracted from the Send path). This mode of operation does not affect the normal operation of other functions.

2.19 Adaptation Control

The adaptation control block regulates the convergence speed of the adaptive filter. When adaptation is disabled by the user the current echo profile is frozen and used to continue cancelling echo. In this state, changes in the echo path are not tracked. Additionally, the Adaptation Control block supports the following:

- Freeze filter adaptation when narrow-band signal is detected
- Freeze filter adaptation when no active far-end speech appears
- Increase convergence speed when active near-end speech is absent
- Decrease convergence speed when low level double talk is present
- Decrease convergence speed further (or freeze adaptation) when high-level double talk is detected