ZL38003



AEC with Noise Reduction & Codecs for Digital Hands-Free Communication

Data Sheet

Zarlink has introduced a new generation family of AEC (ZL38002 and ZL38004). Zarlink recommends these products for new designs.

A full Design Manual is available to qualified customers. To register, please send an email to VoiceProcessing@Zarlink.com.

Features

- · Handles up to -6 dB acoustic echo return loss
- 127 ms acoustic echo canceller
- · Provides up to 12 dB of Noise Reduction
- Operate in two modes, Dual Analog mode and single Analog mode (other port is digital PCM)
- PCM Data Formats in single Port mode- 16-bit Linear, companded ITU-T A-law or U-law
- Advanced NLP design full duplex speech with no switched loss on audio paths
- · Tracks changing echo environment quickly
- Adaptation algorithm converges even during Double-Talk
- Designed for performance in high background noise environments

May 2007

Ordering Information

ZL38003GMG 81 Ball CBGA Trays, Bake & Drypack ZL38003GMG2 81 Ball CBGA**Trays, Bake & Drypack **Pb Free Tin/Silver/Copper

-40°C to +85°C

- Provides protection against narrow-band signal divergence
- Howling prevention eliminates uncontrolled oscillation in high loop gain conditions
- AGC on speaker path
- · Transparent data transfer and mute options
- · Boot loadable for future factory software upgrades
- · Serial micro-controller interface
- Two 16 bit linear ACD and DAC that meet ITU-T G.711/712 recommendations
- Four Audio TX/RX Interfaces
- Differential Microphone Inputs
- Programmable Bias Voltage Output for Electret microphones
- Microphone Presence Detection, Microphone Mute

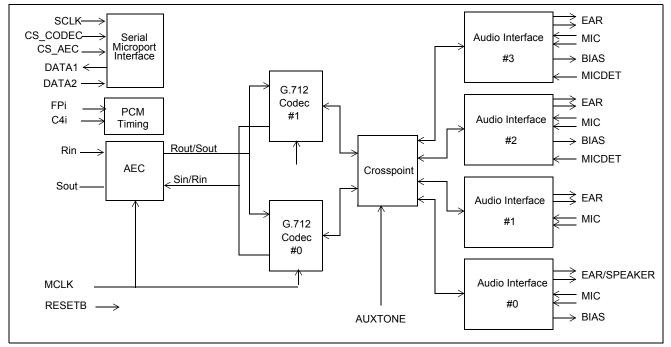


Figure 1 - Block Diagram

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- Multiple Gain pad settings
 - Adjustable gain pads from -24 dB to +21 dB at Xin, Sin and Sout to compensate for different system requirements
 - Programmable Microphone Gain (0dB to +46.5 dB in 1.5 dB Steps)
 - Side tone Mute, Programmable Side tone Gain (-39 dB to +6 dB in 3 dB Steps)
 - User gain control provided for speaker path (-24 dB to +21 dB in 3 dB steps)
 - Programmable Earpiece Gain (-28 dB to +2 dB in 2 dB Steps)
 - RX Channel Mute, Programmable RX Volume control (-21 dB to 0 dB in 3 dB Steps)
- Differential Earpiece Driver Outputs (66 mW rms into 32 Ohms, 150 mW rms into 16 Ohms)
- Cross-Point Connects PCM Channels to any of the Four Audio TX/RX Interfaces

Applications

- Hands-free car kits
- Full duplex speaker-phone for digital telephone
- · Echo cancellation for video conferences
- Intercom Systems
- Security Systems

	MT93L16	ZL38001	ZL38002	ZL38003
Description	AEC for analog hands- free communication	AEC for analog hands- free communication	AEC with noise reduction for digital hands-free communication	AEC with noise reduction & codecs for digital hands-free communication
Application	Analog Desktop phone Analog Intercom	Analog Desktop phone Analog Intercom	Hands-free Car Kits Digital Desktop Phone Home Security Intercom & Pedestals	Hands-free Car Kits Digital Desktop Phone Home Security Intercom & Pedestals
Features				
AEC	1 channel	1 channel	1 channel	1 channel
LEC	1 channel	1 channel	Custom Load	Custom Load
Gains	User Gain	User Gain/18 dB Gain on Sout	User Gain + System tuning gains	User Gain + System tuning gains
Noise Reduction	Ν	Ν	Y	Y
Integrated Codecs	Ν	Ν	Ν	dual channel

1.0 Functional Description

The ZL38003 is an Acoustic Echo Canceller (AEC) with dual codec as shown in Figure 2. The ZL38003 provides 127 ms of acoustic echo cancellation, which makes it ideal for hand free car kits, and speaker phones designs. Each of the codecs in the dual codec can be connected to 1 of 4 four analog ports through a cross point switch. Also, the network side can be routed to a digital PCM interface that input/outputs either linear 2's complement or A-/mu law commanded PCM data.

2.0 Acoustic Echo Canceller (AEC) Description

The AEC section is comprised of an acoustic echo canceller, noise reduction and the operational control functions for operation. The AEC guarantees clear signal transmission in both transmit and receive audio path directions ensuring reliable voice communication even when low level signals are provided. The AEC does not use variable attenuators during double-talk or single-talk periods of speech, as do many other acoustic echo cancellers for speakerphones. Instead, the AEC provides high performance full-duplex operation similar to network echo cancellers. This results in users experiencing clear speech and uninterrupted background signals during the conversation and prevents subjective sound quality problems associated with "noise gating" or "noise contrasting".

The AEC uses an advanced adaptive filter algorithm that is double-talk stable, allowing convergence even while both parties are talking. This algorithm continually tracks changes in the echo path, regardless of double-talk, as long as a reference signal is available for the echo canceller.

The echo tail cancellation capability of the acoustic echo canceller has been sized appropriately (127 ms) to cancel echo in an average sized office or large size car with a reverberation time of less than 127 ms.

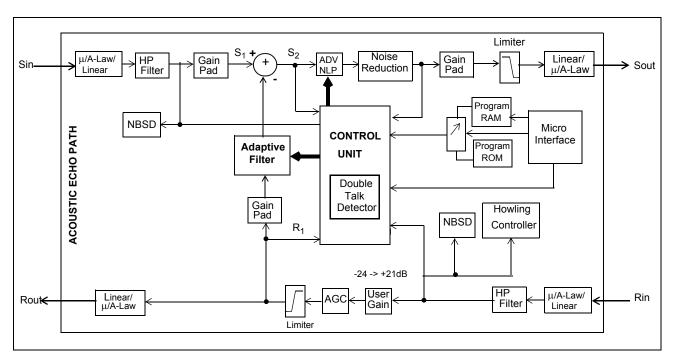


Figure 2 - AEC Block Diagram

2.1 In addition to the echo cancellers, the following functions are supported:

- 12 dB of noise reduction
- User gain control provided for speaker path (-24 dB to +21 dB in 3 dB steps)
- · Gain pads at the Sin and Sout ports plus one at the input of adaptive filter (XRAM)
- Control of adaptive filter convergence speed during periods of double-talk, far end single-talk and near-end echo path changes
- · Control of Non-Linear Processor thresholds for suppression of residual non-linear echo
- · Howling detector to identify when instability is starting to occur and to take action to prevent oscillation
- · Narrow-Band Detector for preventing adaptive filter divergence caused by narrow-band signals
- · Programmable high pass filters at Rin and Sin for removal of DC components in PCM channels
- · Limiters that introduce controlled saturation levels
- · Serial controller interface compatible with Motorola, National and Intel micro controllers
- PCM encoder/decoder compatible with m/A-Law ITU-T G.711, m/A-Law Sign-Mag or linear 2's complement coding
- Automatic gain control on the receive speaker path
- Idle channel noise suppression

3.0 Dual Codec Description

The CODEC Dual Codec provides complete audio to PCM interfaces including filtering and optional data companding as required by the ITU-T G.711 & G.712 recommendations. Programmable gain allows adjustment for a wide range of transducer sensitivities - two microphone amplifiers and four ear piece amplifiers are provided to allow connection to a handset, headset, auxiliary channel and microphone/speaker. A cross-point circuit allows either codec to be connected to any of the four audio interfaces. Programmable voltage sources are available for electret biasing on the Microphone channels.

PCM voice data is passed via a serial interface which operates in ST-BUS or GCI mode. ST-BUS mode allows the Codecs to be allocated to any of the 32 available channels. Control and programming of the Codecs is carried out over a flexible serial micro-controller interface.

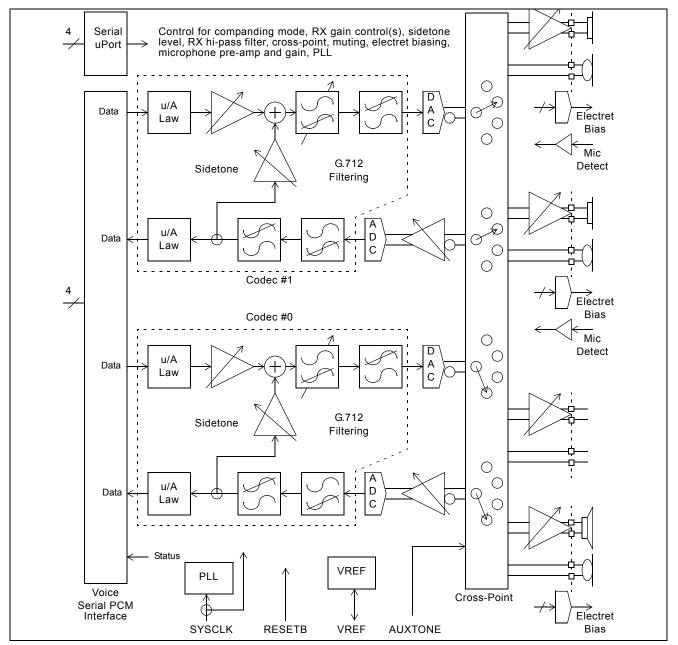
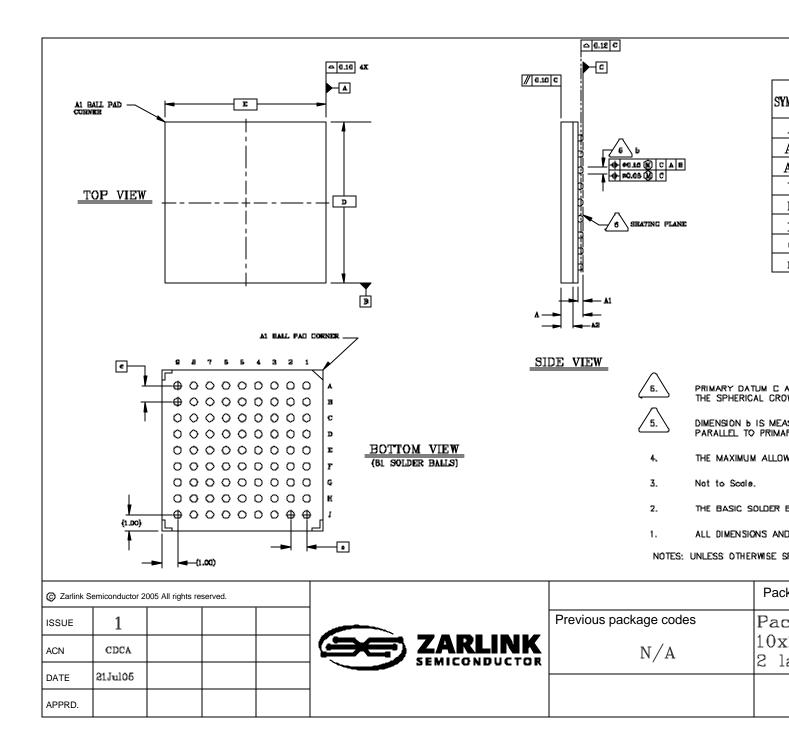


Figure 3 - Block Diagram





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