Acoustic Echo Canceller AEC G4

Wideband and Narrowband Acoustic Echo Cancelation

PRODUCT DESCRIPTION

Acoustic Echo is caused by direct and indirect feedback from speaker to microphone. In order to combat the echo phenomenon, an echo canceller is employed. Today's echo cancellers use sophisticated algorithms running on high speed Digital Signal Processors (DSPs). An acoustic echo canceller (**AEC**) algorithm identifies audio entering the acoustic space, this audio is used as a reference signal in the AEC. The AEC then uses this information to subtract out the transmitted signals that match this "reference" signal from the original audio signal picked up by the microphones. The resulting signal is then transmitted back canceling out the echo.

The Adaptive Digital Technologies acoustic echo canceller electronically removes both direct coupling and reflected echo, enabling true full-duplex hands-free telephony in mobile phones, speakerphones, and hands-free intercoms. Adaptive Digital has been developing echo canceller technology for over ten years. The AEC Wideband/Narrowband (**WB/NB**) is the fourth generation (**G4**) acoustic echo canceller offered by Adaptive Digital.

Speech processing is transitioning to wider bandwidths. The benefits of wideband telephony include increased clarity, and comprehension, and a better overall experience, but also create acoustical challenges. High quality, full-duplex acoustic echo cancellation is an integral component of a hands-free telephony system. Adaptive Digital's AEC WB is a robust Acoustic Echo Canceller which provides true cancellation without tricks or shortcuts that degrade voice quality.

FEATURES

- Superior Double-Talk Performance
- Supports 8 kHz and 16 kHz sampling rates
- Able to achieve greater than 40 dB of ERLE without nonlinear processor
- Supports tail length up to 256 milliseconds
- Integrated noise-reduction
- Integrated automatic gain control Parameters are user configurable
- Non-linear processor
- Excellent voice quality
- Fast Convergence and reconvergence
- No divergence due to doubletalk
- Integrated microphone equalization

WIDEBAND FEATURES

- Full duplex performance under a wide dynamic range of audio levels.
- Supports wideband audio (16 kHz, 32 kHz, 44.1, and 48 kHz sampling rates) with no artificial cutoff of high frequencies.
- Supports tail lengths up to 250 msec. with true full-duplex cancellation. (Not half-duplex suppression.)
- Converges within one second regardless of tail length and sampling rate.

AVAILABILITY

ADT AEC WB/NB is available on the following Platforms: Other configurations are available upon request.

Product	Platform	Memory Model	Endian	Code Gen Tool Version
ADT_aec_G4_c64x	TI TMS320C64x	L3	Little	N/R
ADT_aec_G4_C64x+ / _c66x	TI TMS320C64x+ / C66x	L3	Little	N/R
ADT_aec_G4_ c674x	TI TMS320C674x	L3	Little	N/R
ADT_aec_G4_c55x	TI TMS320C55x	Large	Little	N/R
ADT_aec_G4_armA8	ARM Cortex-A8	N/A	Little	GCC v 4.5.2*
ADT_aec_G4_armA9	ARM Cortex-A9	N/A	Little	GCC v 4.5.2*
ADT_aec_G4_armA15	ARM Cortex-A15	N/A	Little	GCC v 4.5.2*
ADT_aec_G4_win32dll	Windows	N/A	Little	N/R
ADT_aec_G4_win32lib	Windows	N/A	Little	N/R

*GCC v 4.5.2 (Sourcery G++2011.03-41)

Endian, byte order: “Little Endian” means that the low-order byte of the number is stored in memory at the lowest address, and the high-order byte at the highest address. "Big Endian" means that the high-order byte of the number is stored in memory at the lowest address, and the low-order byte at the highest address.

Acronyms

Mm – Memory Model: Memory Model is specific to Texas Instruments processors.

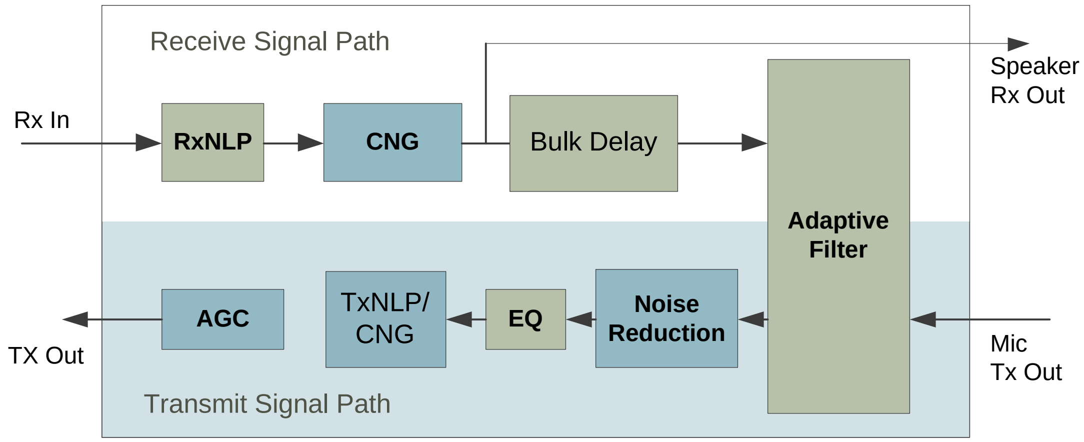
N/A – Not Applicable

N/R – Not Recorded

FUNTIONAL DESCRIPTION

The following figure is a simplified block diagram of the Acoustic Echo Canceller. A description follows. In the description, you will find parameter names underlined and in inside parenthesis (parameter). This indicates that a user-controlled parameter is involved in the process being described.

The top half of the diagram shows the receive signal path, or the signal path from the telephone network to the speaker. The bottom half of the diagram shows the transmit signal path from the microphone toward the telephone network. The AEC cancels the echo that occurs between the speaker output and the microphone input.



The terms Rx (Receive) and Tx (Transmit) may be confusing at first because both the receive and transmit paths have inputs and outputs. The names receive and transmit are used from the point-of-view of the person at the speaker/microphone side.

The RxIn signal coming from the network is fed into the RxNLP (Receive Nonlinear Processor). The RxNLP can attenuate the received signal by a variable amount based upon the talk state (single talk vs. double-talk). This attenuation improves the overall echo attenuation.

The output of the RxNLP is fed both to the transmit output (TxOut) and into the bulk delay block, whose delay is contolled by bulkDelaySamples. The bulk delay block compensates for the buffering delay at the RxOut and TxIn interfaces as well as any other non-acoustic system delays in the path between RxOut and TxIn. The output of the bulk delay is fed to the the adaptive filter.

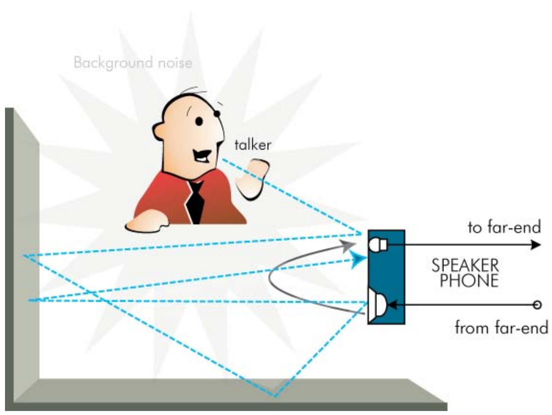
The adaptive filter estimates the echo and subtracts it from the TxIn signal to form the residual signal. The residual signal is fed to the noise reduction block. This noise reduction block removes background noise and therefore improves the signal to noise ratio of the transmit signal.

The output of the noise reduction block is fed into an equalizer. The equalizer is used to flatten out the frequency response of the transmit channel. This may be necessary due to the acoustics of the hands-free device and due to the characteristics of the microphone itself. The output of the transmit equalizer is fed into the TxNLP. The TxNLP is the transmit non-linear processor. The TxNLP increases the echo attenuation by attenuating the residual by a variable amount based upon the talk state.. The TxNLP block also includes a comfort noise generator.

The compute gain block computes the AGC gain. The output of the TxNLP is fed into the AGC gain block, which provides gain or loss depending upon the residual signal level. The output of the AGC is fed to the TxOut output of the AEC.

USER CONTROLLED PARAMETERS (SUMMARY)

- Sampling Rate
- Tail Length
- Frame Size
- NLP Control
- AGC Control
- Equalizer Control
- Noise Reduction Control



SPECIFICATIONS

The following tables contains CPU utilization of the AEC as a function of sampling rate, tail length, and processor. The CPU utilization is given in Millions of Instruction Cycles Per Second per channel. This is also referred to as MIPS, which we also equate to processor clock speed specified in MHz.

TI TMS320

C6000x

CPU UTILIZATION & MEMORY REQUIREMENTS

All Memory usage is given in units of byte.

Sampling Rate (Hz)	Tail Length (msec)	MIPS	Program Memory			Data	Scratch	Per-channel	Stack
			C64x	C64x+	C674x				
8000	32	26	81K	69K	62K	2K	0	26K	20K
8000	64	32				2K	0	41K	20K
16000	32	68				2K	0	41K	20K
16000	64	85				2K	0	77K	20K

Specifications are approximate. Exact per-channel numbers can be obtained by using API functions AECG4_alloc or AECG4_staticAllocHelper.

C55x

CPU UTILIZATION & MEMORY REQUIREMENTS

The following table contains CPU utilization of the AEC as a function of sampling rate, tail length, and processor. The CPU utilization is given in Millions of Instruction Cycles Per Second per channel. This is also referred to as MIPS, which we also equate to processor clock speed specified in MHz.
All Memory usage is given in units of byte.

Sampling Rate (Hz)	Tail Length (msec)	MIPS	Program Memory	Data	Scratch	Per-channel	Stack
8000	32	30	46K	6K	0	55K	20K
16000	32	65		6K	0	63K	20K

MIPS measured on 5510 DSK, AGC, NR and CNG were enabled. NR2 is not on.

Specifications are approximate. Exact per-channel numbers can be obtained by using API functions AECG4_alloc or AECG4_staticAllocHelper.

ARM® DEVICES

CORTEX-A8/A9/A15

CPU UTILIZATION & MEMORY REQUIREMENTS

All Memory usage is given in units of byte.

Sampling Rate (Hz)	Tail Length (msec)	MIPS	Program	Data	Scratch	Per Channel	Stack
8000	32	31.5	113K	6.7K	0	26K	20K
8000	64	38				41K	20K
16000	32	71.5				41K	20K
16000	64	94				77K	20K

Specifications are approximate. Exact per-channel numbers can be obtained by using API functions AECG4_alloc or AECG4_staticAllocHelper.

PC/Windows

Win32

CPU UTILIZATION & MEMORY REQUIREMENTS

All Memory usage is given in units of byte.

Sampling Rate (Hz)	Tail Length (msec)	MIPS	Program Memory	Data	Scratch	Per-channel	Stack
8000	32	51	61K	2K	0	26K	20K
8000	64	68				41K	20K
16000	32	113				41K	20K
16000	64	136				77K	20K

Specifications are approximate. Exact per-channel numbers can be obtained by using API functions AECG4_alloc or AECG4_staticAllocHelper.

FUNCTIONS

`AECG4_ADT_create(…)`	Create and initializes an echo canceller channel
`AECG4_ADT_apply(…)`	Executes cancellation function
`AECG4_ADT_backgroundHandler(…)`	Handles background calculations
`AECG4_ADT_delete(…)`	Deletes an echo canceller channel

Deliverables

The deliverable items are platform dependent. In general, there is one library. (Sometimes multiple variants of the library are included in the deliverables.) There are also header files, some of which are specific to the product and others are common across many of Adaptive Digital's products. Also included in the deliverables is product documentation, which includes a users guide and usually includes release notes and a data sheet. Sample/test code may be included as well.

Adaptive Digital is a member of the Texas Instruments Developer Network, and ARM Connected Community.

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