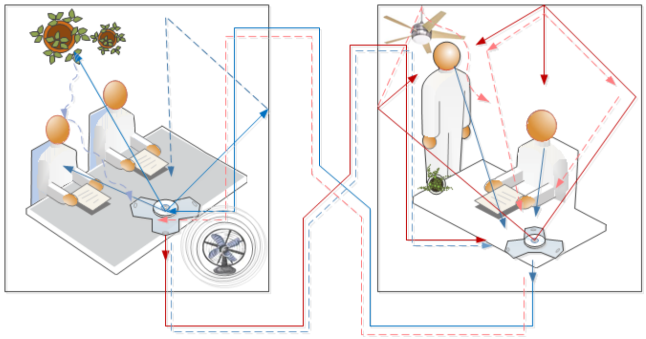
HD AEC™

High Definition Acoustic Echo Cancellation

Adaptive Digital's HD AEC™ is a High Definition (HD), Multi-Mic Capable, Full-Duplex Acoustic Echo Canceller (AEC) which includes noise reduction (NR), as well as anti-howling, adaptive filtering, nonlinear processing, and double-talk detection. It operates in narrowband (NB), wideband (WB), super-wideband (SWB) and full band (FB) frequency ranges.



Adaptive Digital's HD AEC product provides superior voice clarity and true full duplex performance under a wide set of challenging acoustic environments. It is capable of eliminating the acoustic echo in difficult conditions such as unbalanced speech levels, close speaker to mic proximity, indoor/outdoor environments, background noise, reflective room surface, double talk, and echo path changes.

Adaptive Digital's HD AEC technology can be found in varied markets including High-end to Mid-level Corporate Conference centers, Smart Home Control/Automation, High-end Intercom Systems (Door Bells), Healthcare, and Home/Baby monitoring systems and essentially anywhere where voice quality is affected by adverse room conditions. Additionally, the HD AEC is effective in improving the performance of speech recognition algorithms when operating in an echoic environment

Available on Android, iOS, ARM®v7 (Cortex-A8, A9, A12, A15, A17), ARMv8 (Cortex-A53, A57) TI TMS320C6000™, MIPS32®, Window/x86, Linux/x86 Supports the following Operating Systems (OS): iOS 32/64 bit, Linux 32/64 bit, Android 32/64 bit, Windows

FEATURES

- True full-duplex operation under a wide dynamic range of audio levels, even when microphone input signal is weak
- Programmable sampling rate, supporting **narrowband** (8 kHz), **wideband** (16 kHz), **super-wideband (32 kHz), and full-band (44.1, 48 kHz).**
- **Improved adaptive nonlinear processor**
- **Handles echo tails of up to 400 msec. and greater,** with true full-duplex cancellation. (Not half-duplex suppression.)
- Spectrally representative comfort noise generator.
- Automatically adjusts for unknown bulk (buffering/audio driver) delay
- Able to handle strong echo (speaker to microphone gains up to 20 dB)
- **Anti-Howling**
- Instantly adjusts to user-controlled speaker gain changes
- Handles external user-controlled volume changes
- Parameters are user configurable
- Improved fast convergence and reconvergence
- No divergence due to doubletalk
- Integrated Automatic Gain Control
- Improves speech recognition performance in an echoic environment.
- Integrated Next Gen Noise reduction
- Integrated Transmit Equalization

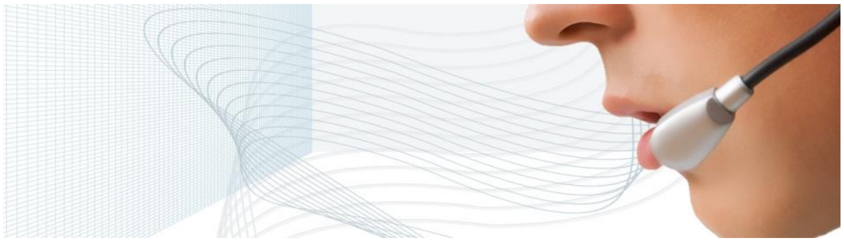
AVAILABILITY

Adaptive Digital's HD AEC is available on the following Platform(s): Other configurations are available upon request.

Platform
ARM Cortex-A8, A9, A12, A15, A17, A53, A57
ARM Cortex-M4
TI TMS320C6000 - C64x, C64x+, C674x | TMS320C5000 – C55x
Windows (DLL or Static Library)
Linux/x86
MIPS32 - 24KEc

Can be built with customer specific tool chain.

PRODUCT DESCRIPTION



The Adaptive Digital Technologies high definition acoustic echo canceller (HD AEC), has integrated Noise Reduction and AGC into its AEC algorithm, and created appropriate hooks to make them work together seamlessly.

Noise Reduction is done pre-NLP, resulting in a far cleaner audio stream feeding into the non-linear processor. By making the AGC aware of the AEC state, we can avoid having the AGC becoming a cause of howling. Changes in gain can adversely affect an AEC; Adaptive Digital's HD AEC has the ability to adapt to changes in the acoustic path (including gain/loss changes.) And when the changes are known, like in the case of controlled gain changes, Adaptive Digital's HD AEC has hooks that enable the application to tell it the nature of the gain change so it can adjust immediately rather than take time to reconverge.

Howling can occur when there is a full-duplex communication link that has echo at both ends. These echo, or coupling, paths create feedback loops. In full-duplex communication systems where, by definition, both communication paths are open at all time, howling can be a serious issue. With Anti-howling enabled the HD AEC identifies when instability is starting to occur and takes action to mitigate the instance of feedback looping. HD AEC electronically removes both direct coupling and reflected echo, enabling true full-duplex hands-free telephony.

SPECIFICATIONS

NB - narrowband (8 kHz), WB - wideband (16 kHz), SWB - super-wideband (32 kHz), and FB - full-band (44.1, 48 kHz).

All Memory usage is given in units of byte. Specifications are approximate. Exact per-channel numbers can be obtained by using API functions HDAEC_alloc or HDAEC_staticAllocHelper.

ARM® DEVICES

ARM CORTEX V7A UPWARD COMPATIBLE TO **ARMV8A** (ARM CORTEX-A53/57)

HD AEC CPU UTILIZATION & MEMORY REQUIREMENTS – ARM Cortex-A8/A9/A12/A15/A17

Platform	Sampling Rate	Tail Length (msec)	MIPS*per Mic	Per Channel Memory
ARM Cortex v7A	8 kHz	32	25.75	24k
		64	29.75	33k
		128	36.5	54k
		256	51.25	109k
		320	388	143k
		400	44.75	191k
ARM Cortex v7A	16 kHz	32	59	44k
		64	60.5	61k
		128	76.5	96k
		256	112	180k
		320	147.5	228k
		400	176	293k
ARM Cortex v7A	**32 kHz**	32	119	61K
		64	153	96K
		128	226	180K
		256	325	396K
		320	449	529K
		400	595	718K
ARM Cortex v7A	**48 kHz**	32	201	78K
		64	277.5	136K
		128	519	280K
		256	886.5	678K
		320	1087.5	932K
		400	1428	1302K

*with anti-howling

Note: **MIPS generated with single mic enabled**, and running with on-chip (internal) program and data memory only.

ARM CORTEX -M4

HD AEC CPU UTILIZATION & MEMORY REQUIREMENTS - ARM

Platform	Sampling Rate	Tail Length (msec)	MIPS*per Mic	Per Channel Memory
ARM Cortex-M4	8 kHz	32	25.75	24k
		64	29.75	33k
		128	36.5	54k
		256	51.25	109k
		320	388	143k
ARM Cortex-M4	16 kHz	32	59	44k
		64	60.5	61k
		128	76.5	96k
		256	112	180k
		320	147.5	228k
ARM Cortex -M4	**32 kHz**	32	119	61K
		64	153	96K
		128	226	180K
		256	325	396K
		320	449	529K
ARM Cortex -M4	**48 kHz**	32	201	78K
		64	277.5	136K
		128	519	280K
		256	886.5	678K
		320	1087.5	932K

Note: HD AEC Cortex-M4 MIPS generated with **0 wait state FLASH.**

Specifications measure on TI Tiva C series ARM Cortex-M4 based MCU.

TEXAS INSTRUMENTS

TEXAS INSTRUMENTS TMS320C**6000 – C674X**

HD AEC CPU UTILIZATION & MEMORY REQUIREMENTS – C674X

Platform	Sampling Rate	Tail Length (msec)	MIPS*per Mic	Per Channel Memory-byte
C674x	8 kHz NB	32	19	25K
		64	20	34K
		128	21	55K
		256	22	110K
		320	24	143K
		400	62	160K
C674x	16 kHz WB	32	33	45K
		64	34	61K
		128	36	96K
		256	38	180K
		320	43	223K
		400	48	250K
C674x	**32 kHz SWB**	32	65	56K
		64	67	86K
		128	71	157K
		256	78	350K
		320	81	470K
		400	85	600K
C674x	**48 kHz FB**	32	78	68K
		64	103	114K
		128	111	234K
		256	126	584K
		320	133	814K
		400	141	1100K

TEXAS INSTRUMENTS TMS320C**6000 – C64X+**

HD AEC CPU UTILIZATION & MEMORY REQUIREMENTS – C64X+

Platform	Sampling Rate	Tail Length (msec)	MIPS*	Per Channel Memory-byte
C64x+	8 kHz NB	32	22	30K
		64	27	45K
		128	35	81K
		256	51	179K
		320	57	240K
		400	63	300K
C64x+	16 kHz WB	32	43	54K
		64	51	80K
		128	67	140K
		256	101	282K
		320	119	365K
		400	150	450K
C64x+	**32 kHz SWB**	32	85	75K
		64	103	128K
		128	136	259K
		256	204	620K
		320	238	847K
		400	300	1007K
C64x+	**48 kHz FB**	32	112	99K
		64	134	185K
		128	190	410K
		256	292	1085K
		320	344	1534K
		400	399	1997K

*with anti-howling

Note: **MIPS generated with single mic enabled**, and running with on-chip (internal) program and data memory only. When using external source for program and data memory, MIPS increase by 3x per enabled microphone.

TEXAS INSTRUMENTS TMS320C**5000 – C55X**

HD AEC CPU UTILIZATION & MEMORY REQUIREMENTS - ARM

Platform	Sampling Rate	Tail Length (msec)	MIPS	Program Memory	Data	Per-channel
C5510	8 kHz	32	32	46K	6K	20K
C5510	16 kHz	32	65			20K

Note: MIPS generated with AGC, NR and CNG enabled. NR2 is not turned on.

PC/WINDOWS

X86

HD AEC CPU UTILIZATION & MEMORY REQUIREMENTS – Windows/x86

Platform	Sampling Rate	Tail Length (msec)	MIPS*per Mic	Per Channel Memory -byte
Windows/x86	8 kHz NB	32	81	28K
		64	92	37K
		128	127	58K
		256	186	113K
		320	220	147K
		400	262	195K
Windows/x86	16 kHz WB	32	160	52K
		64	194	68K
		128	248	104K
		256	373	187K
		320	429	235K
		400	506	301K
Windows/x86	**32 kHz SWB**	32	821	56K
		64	1089	86K
		128	1409	157K
		256	2284	350K
		320	2489	470K
		400	2702	600K
Windows/x86	**48 kHz FB**	32	1266	68K
		64	1526	114K
		128	1913	234K
		256	2966	584K
		320	3493	814K
		400	4100	1004K

*with anti-howling

Note: **MIPS generated with single mic enabled**, and running with on-chip (internal) program and data memory only.

MIPS® IMAGINATION TECHNOLOGIES

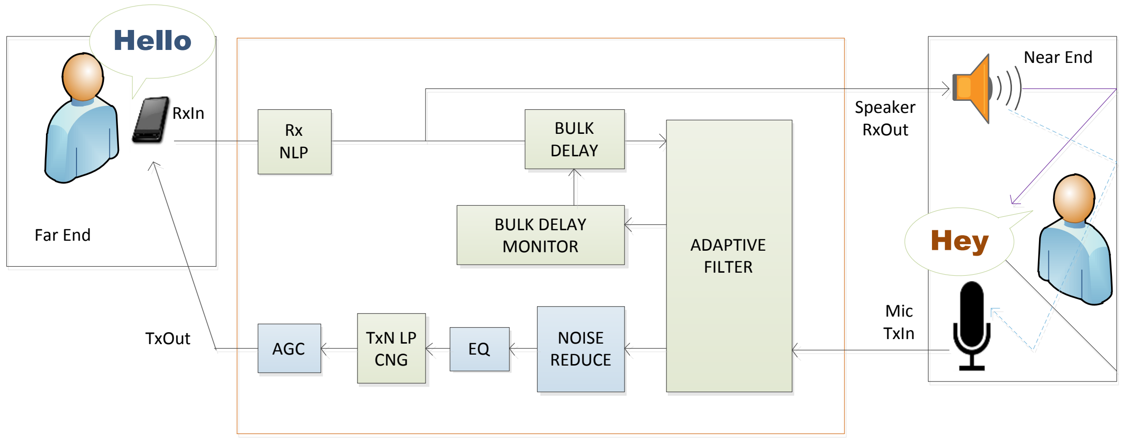
MIPS32® – 24KEc®

The 24KEc core integrates the MIPS® DSP Application-Specific Extension (ASE). The 24KEc is a 32-bit RISC core for high performance applications. Contact sales at 1-800-340-2066 for additional information.

FUNCTIONAL DESCRIPTION

The following figure is a simplified block diagram of the Acoustic Echo Canceller. A description follows.

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 HD AEC cancels the echo that occurs between the speaker output and the microphone input.



AGC – Auto Gain Control	CNG – Comfort Noise Generator	Rx - Receive
NLP –Non-Linear Processor (enhanced)	EQ – Equalizer	Tx – Transmit

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). Under difficult acoustic conditions, the RxNLP can improve full-duplex operation and hence the overall voice quality.

The output of the RxNLP is fed both to the transmit output (TxOut) and into the bulk delay block. 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 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 adaptive filter works in conjunction with the bulk delay monitor, which monitors and adjusts bulk delay in situations where the bulk delay is unknown due non-deterministic audio drivers.

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 transmit non-linear processor (TxNLP). 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.

Automatic Gain Control (AGC) is provided to help boost lower level speech signals in hands-free environments. 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.

In the multi-microphone case, there is still a single receive path but there is one transmit path per microphone.

In the case of multi-microphone noise reduction, there is a single receive path, a complete transmit path for the primary microphone, and a partial transmit path for the secondary microphone. In this case, there are two transmit inputs (one for each microphone) but only one transmit output containing the echo cancelled and noise reduced signal.

USER CONTROLLED PARAMETERS (SUMMARY)

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

PERFORMANCE

ERLE as a function of echo tail length

As the echo tail becomes longer, background noise and computational error can reduce the ERLE. So we also provide some data on ERLE vs. Tail Length.

- ERLE – Echo Return Loss Enhancement is a measure of the amount of echo attenuation an AEC can achieve without the use of nonlinear processing.

ERLE vs. Active Tail Length – 8 kHz sampling rate (at ERL = 0 dB)

Active Tail Length	ERLE
32	38
64	38
128	37
256	37
512	36

ERLE vs. Active Tail Length – 16 kHz sampling rate (at ERL = 0 dB)

Active Tail Length	ERLE
32	32
64	33
128	35
256	35
512	19.9

ERLE vs. Active Tail Length – 48 kHz sampling rate (at ERL = 0 dB)

Active Tail Length	ERLE
32	34
64	33
128	32
256	29

FUNCTION APIs

```
HDAEC_
      AEC_ADT_create(..)            Create and initializes an echo canceller
      AEC_ADT_createMMIC(…)         channel (single mic, multi-mic, dual-mic
      AEC_ADT_createDMNR(…)         with noise reduction)

      AEC_ADT_apply(…)              Executes cancellation function
      AEC_ADT_applyMMIC(…)

      AEC_ADT_applyTx(…)            Executes Transmit Only (for applications
                                    that require split tx/rx processing
      AEC_ADT_applyRx(…)            Executes Receive Only (for applications
                                    that require split tx/rx processing
      AEC_ADT_control(…)            Modify AEC parameters. Obtain status.
      AEC_ADT_controlMMIC(…)

      AEC_ADT_delete(…)             Deletes an echo canceller channel
      AEC_ADT_deleteMMIC(…)
```

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.

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