ASR PREPROCESSOR SOFTWARE

Adaptive Digital Technologies, Inc.

AUTO SPEECH RECOGNITION (ASR) PREPROCESSOR

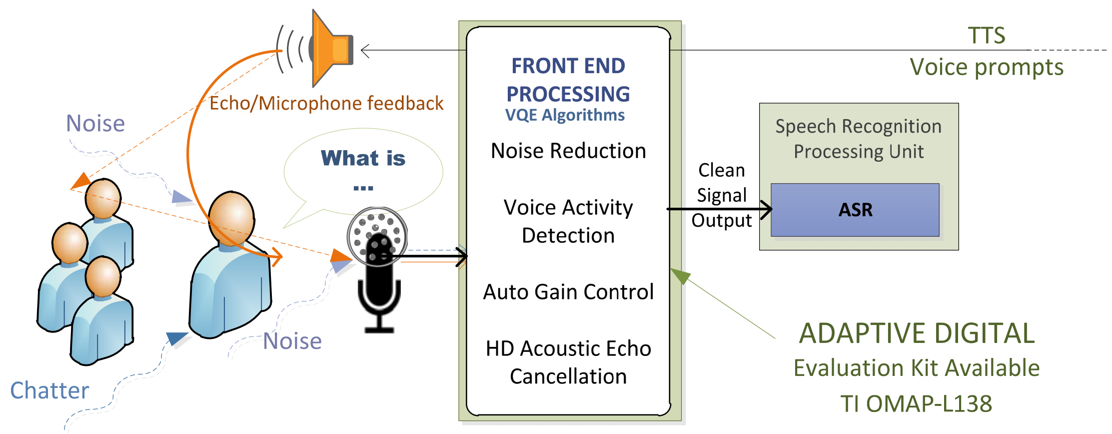
ASR Preprocessor Suite, speech recognition enhancement application software

Speech recognition performance degrades drastically under noisy and reverberant environments. The mixing of background noise with the speech of interest results in a dramatic decline of speech recognition accuracy in the presence of noise and reverberation. This is especially true when the background noise is itself speech. The effect worsens as the distance between the talker and the microphone increases.

Adaptive Digital's **ASR Preprocessor** encompasses an integrated set of front-end signal processing Voice Quality Enhancement (VQE) features to extract the relevant speech signal, and applies techniques to reduce the noise, cancel echoic interference, missing data, and emphasize the desired signal.

FEATURES

- High Definition Acoustic Echo Cancellation (HD AEC)
- Voice Activity Detection (VAD)detects the presence of speech
- Robust Background Noise removal
- Automatic Gain Control (AGC)
- Auto Level Controls
- Packet Loss Concealment (PLC)*(ASR in VoIP Environment)*



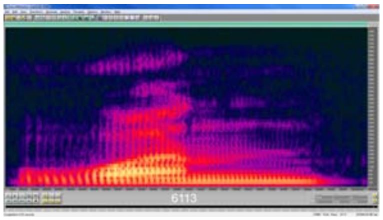
BENEFITS

- The ASR software includes Adaptive Digital's proprietary **High Definition Acoustic Echo cancellation** (HDAEC). *HDAEC has configuration settings that customize for use with ASR.*

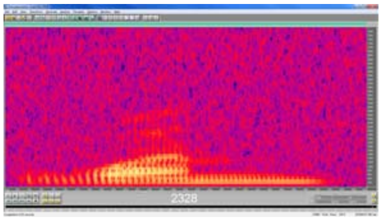
- **Superior Noise Reduction** specifically designed for use in conjunction with ASR to remove noise and leave the important speech of interest signal intact.

- Supports narrowband (8 kHz), wideband (16 kHz), **super-wideband (32 kHz), and full-band (44.1, 48 kHz).**

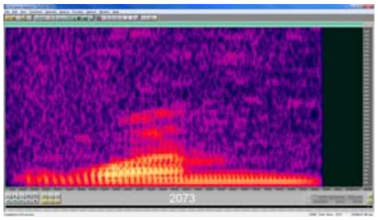
ASR PREPROCESSING SPECTROGRAM COMPARISON



Original Speech



Speech with noise



After ASR Preprocessor

TEST RESULTS

*HD ACOUSTIC ECHO CANCELLATION WITH IMPAIRMENT**

Talker Distance (inches)	Far End Level	Reverb Time	Noise level	Recognition Rate Without Enhancement	Recognition Rate With AEC Enhancement
2	-10	256	-96	50.00%	100.00%
12	-10	256	-96	5.00%	100.00%
24	-10	256	-96	0.00%	96.00%
36	-10	256	-96	0.00%	94.00%
42	-10	256	-96	0.00%	81.00%
48	-10	256	-96	0.00%	63.00%
2	-10	256	-25	63.00%	100.00%
12	-10	256	-25	0.00%	100.00%
24	-10	256	-25	0.00%	98.00%
36	-10	256	-25	0.00%	83.00%
42	-10	256	-25	0.00%	76.00%
48	-10	256	-25	0.00%	43.00%

VQE Algorithms – HD AEC, Noise Reduction, AGC

* ASR Preprocessor performance in a situation where there is full-duplex voice, background noise, and reverberation.

TALKER DISTANCE WITH IMPAIRMENT

Talker Distance (inches)	Reverb Time (msec)	Recognition Rate Without Enhancement	Recognition Rate With Enhancement
2	400	100.00%	**100.00%**
12	400	100.00%	**100.00%**
24	400	100.00%	**100.00%**
36	400	96.00%	**100.00%**
42	400	94.00%	**100.00%**
48	400	47.00%	**76.00%**

VQE Algorithms – Noise Reduction, AGC

SIGNAL TO NOISE RATIO - DB

Signal To Noise	Recognition Rate Without Enhancement	Recognition Rate With Enhancement
18	100	100
16	100	100
14	100	100
12*	54	100
10*	72	100
8	61	85

VQE Algorithms – Noise Reduction, AGC

`* Recognizer gave up part way through without ASR preprocessor enabled.`

DESCRIPTION

Speech recognition systems have a high level of accuracy in quiet conditions but work poorly under noisy conditions and are particularly challenged when the distance between the talker and microphone is increased. The accuracy of a speech recognition system may be acceptable if talking into an ASR enabled unit in your quiet office, yet the same unit's performance may be unacceptable in a shopping mall.

What is needed is a more effective method of enhancing the speech of interest for accuracy purpose especially in situations when the ASR unit is positioned far from the speaker in noisy environments including background conversations and multi-person chatter.

Adaptive Digital's ASR Preprocessor Software addresses such degradation issues. The field hardened HD AEC is exceptional in handling acoustic echo. The Noise reduction algorithm has been developed to achieve up to 12 dB of signal to noise ratio improvement with little to no degradation to the desired speech signal. Automatic Level Control, included, has the ability to adjust speech levels speech levels to increase the dynamic range of the ASR, especially at the low signal level end.

Optionally, in VoIP environments Packet Loss Concealment may be required to handle packet drop-outs that would otherwise be perceived as missing data.

The key to integrating a superior solution in an ASR environment is to put various enhancement algorithms together in such a way that maximizes speech quality. Adaptive Digital provides Preprocessor Speech Enhancement algorithms to significantly improve the robustness and accuracy of a speech recognition system.

Adaptive Digital's engineering team has over thirty years' experience in practical and theoretical aspects of DSP Software, and Communications. We support customers from concept through deployment, helping to ensure that our products' voice quality is carried into the end user experience

CUSTOMIZATION/SUPPORT

Customization is often very important to equipment manufacturers who wish to differentiate their products by incorporating features that are not available in competing products. Should modifications be necessary for your project, Adaptive Digital can customize a solution to fit your specific requirements. *Adaptive Digital provides support throughout all project phases; customers have **direct access** to our technical team over the phone or via email.*

AVAILABILITY

Adaptive Digital's ASR Preprocessor algorithms run on a variety of processors and platforms, including Android, iOS, Texas Instruments' TI TMS320C6000™, ARMv7 Cortex-A(A8, A9, A12, A15, A17), ARMv7 Cortex-M4, ARMv8 (Cortex-A53, A57) Window/x86, Linux/x86: Other configurations are available upon request.

Evaluation Kit: Adaptive Digital's ASR Preprocessing Software is available as an Eval Kit on Texas Instruments'' OMAP – L138 C6000 DSP + ARM Processor

The OMAP-L138 C6000 DSP+ARM processor is a low-power applications processor based on an ARM926EJ-S and a C674x DSP core.

Contact Adaptive Digital to discuss your requirements.

ADAPTIVE DIGITAL TECHNOLOGIES

Adaptive Digital has extensive know-how and a proven track record in the telecommunication market, serving more than 100 renowned customers in over 50 countries.

Our algorithms process voice signals every minute of every day in wireless, VoIP, traditional telephony, server, gateway, military, medical and conferencing applications.

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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ARM CONNECTED

TEXAS INSTRUMENTS
TI Developer Network

IMPORTANT NOTICE: Data subject to change, for the most up to date information visit our website. Customers are advised to obtain the most current and complete information about Adaptive Digital products and services before placing orders.

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

