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VoiceBoost™ Real Time Speech Discrimination in Noise

Real time audio signal processing for ultra-low power processors increases speech clarity in noise

The Challenge

Poor understanding of speech amid difficult background noise is among the top reasons for user dissatisfaction with hearing and communication devices. Mobile devices are increasingly used with hands free voice interfaces, while users expect high voice call quality in noisy environments. Poor performance in noise is also one of the reasons that many users of digital hearing aids remove their hearing aids in restaurants and other noisy environments, and is a reason cited for almost half of hearing aid product returns.

Poor speech understanding in noise is due to the fact that existing real-time noise reduction approaches attempt to filter out undesirable background noise by classifying sounds as either speech or noise based on their spectral frequencies. When the background noise is "static" such as fan, wind or vehicle noise with frequencies outside the range of human speech, this approach works well. However when the background noise is "nonstationary" and has the same frequencies as the speech of interest, such as the background babble in a busy airport or a lively restaurant, frequency-based noise reduction approaches cannot separate

the background noise from the speech of interest, resulting in degraded sound quality and lower comprehension by the listener.

Advances in mobile devices and mobile communications networks have led to different methods of interaction which also reduce the effectiveness of noise reduction methods. Even in mobile devices which include a rear-facing microphone to attempt to separate background noise from the speaker's voice picked up by the front-facing microphone, this attempt at noise reduction fails in cases in which people no longer hold the device up to the ear in the traditional "phone" manner. Voice command user interfaces, video calling, tablets, webcams, and hands-free use while driving all tend to move the microphone away from the mouth to arm's length or farther.

These "arms-length" interaction methods allow background noise to mix with the speakers voice, effectively defeating existing two-microphone noise reduction solutions, especially when the background noise is itself speech. What is needed is a more effective method of discriminating the speech of interest from all background noise, including noise with the same frequencies as speech such as the babble of other speakers.

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Product Overview

Malaspina's VoiceBoost™ Real Time Speech Discrimination software isolates the primary speech-of-interest from background noise within audio signals using a patent-pending mathematical model of human speech. VoiceBoost™ Real Time Speech Discrimination is a compact and CPU-efficient implementation of Model Based Speech Discrimination offered as fixed-point C-language software with complete architecture, porting, and functional source code documentation.

VoiceBoost™ Real Time Speech Discrimination is designed for OEMs. It is easily integrated into existing audio processing pipelines for ultra-low power processors used within hearing aids, mobile phones, Bluetooth* headsets and portable two-way radios. It is effective in isolating speech of interest from noisy audio input signals using either a single omni-directional microphone or directional microphone, or using input signals from multiple microphones (beam steering) with spatial separation as small as 5 millimeters (0.2 inches).

Key Features

- Isolates speech of interest from background speech, static noise and impulse noise
- Supports narrowband 8kHz and wideband 16kHz audio
- Speaker and language independent
- Effective with near-field and arms-length audio input signals
- Effective with single or multiple omni-directional or directional microphones
- Supports signal beamforming
- Supports rapid speaker changes and multiple speakers of interest
- Portable fixed-point ANSI 'C' implementation

Applications

- Mobile phones & tablets: arms-length calling or video calls, ASR assist, voice command interfaces
- Digital hearing aids: urban or outdoor speech-mode programs
- Bluetooth* headsets: urban or vehicular environments
- Two-way radio: industrial, first responder or air traffic communications

Performance

Unlike competing speech processing implementations, Malaspina's VoiceBoost™ Real Time Speech Discrimination does not require two or more microphones, large processing power or lengthy signal latencies to separate speech of interest from background noise, including speech-shaped noise and speech babble.

Improvements in speech quality greater than 36% have been shown when VoiceBoost™ speech discrimination is compared to mobile phones and 3G network operators' existing noise reduction algorithms and narrowband codecs.

Malaspina's VoiceBoost™ Real Time Speech Discrimination can be compiled in several configurations to meet varying performance, resource or operational design constraints.

Effective configurations can operate with wideband audio in as little as 8.1 Mcycles and 12ms real time delay on DSPs based on NXP CoolFlux* CF6-24 ultra-low power processor cores, with runtime footprint less than 9Kwords of data memory.
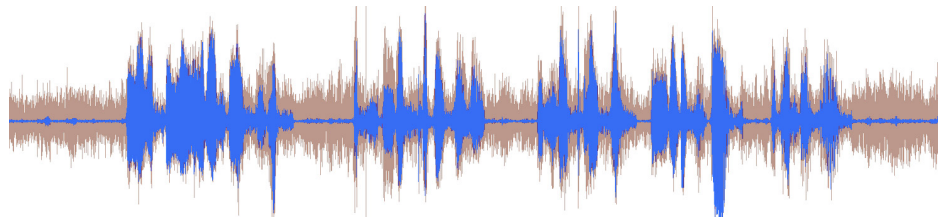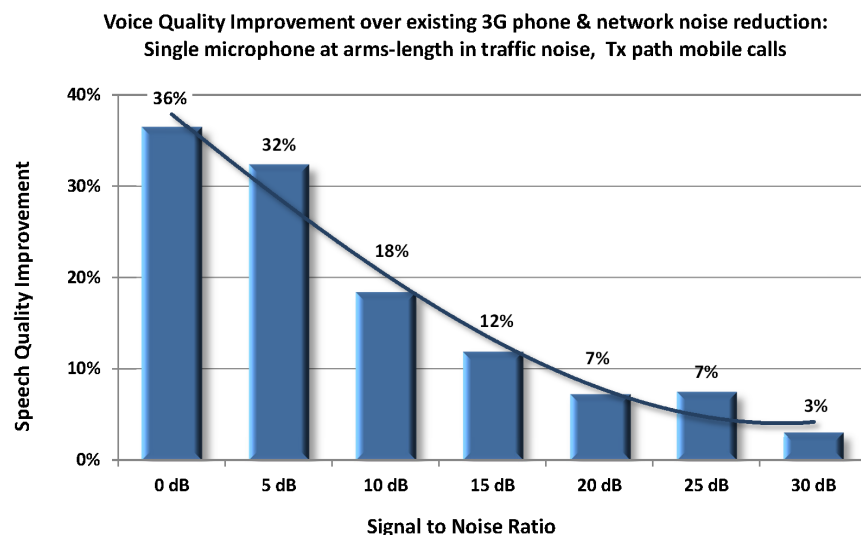
VoiceBoost™ extracted speech signal (shown in blue) from a 5dB SNR signal obtained from a single omnidirectional microphone in 9.3 Mcycles with 12ms real time delay

Figure 1 shows a 25dB signal to noise improvement in an audio signal achieved using a single omni-directional microphone and 12 millisecond real time delay.

Input signals consisted of speech at arm's length from a single omnidirectional microphone, amid varying levels of background noise consisting of omnidirectional babble (background speech) and impulse noise from restaurant environments (clanging dishes) and street environments (traffic noise).

Software Framework

VoiceBoost™ Real Time Speech Discrimination does not impose any restriction upon your selection of operating environment (nor is the presence of an operating system even required). No particular software task management or inter-task communication mechanism is imposed. Malaspina Labs' technologies are designed to be installed as a step of the audio processing pipeline and inter-operate seamlessly with other processing such as band pass filtering, feedback cancellation, equalization and amplification.

Voice Quality Improvement over existing 3G phone & network noise reduction: Single microphone at arms-length in traffic noise, Tx path mobile calls

Speech Quality Improvement (vertical axis)

Signal to Noise Ratio (horizontal axis)

Signal to Noise Ratio	Speech Quality Improvement
0 dB	36%
5 dB	32%
10 dB	18%
15 dB	12%
20 dB	7%
25 dB	7%
30 dB	3%

Application Kits & Custom Solutions

To learn more about VoiceBoost™ Real Time Speech Discrimination and how it can help increase user satisfaction of your communications products in noise, contact Malaspina Labs or a Malaspina Labs channel partner today.

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About Malaspina Labs

Malaspina Labs performs applied research in the field of speech processing. Malaspina and its subsidiaries provide portable software implementations of proprietary algorithms which execute in real-time on ultra-low power processors.