DTEC

AudioFocus

Product Information

Improve the overall quality of your audio signals for more confortable understanding by human operators or for maximizing the accuracy of all kind of speech recognition and verification engines. Based on state of the art AI audio restoration algorithms, ***AudioFocus*** can automatically **remove the most usual background noises** from a speech signal or be **custom trained** for dealing with specific noises.

The solution is based on deep learning models that have been trained for separating a human voice from all kind of backgrounds noises, both periodic (like a running engine or an electric hum) and non-periodic (like background chatter, music, microphone distortion/saturation, bumps, horns and sirens). This feature makes it ideal for improving the perceived quality of **real time communications** or for **batch processing** of large quantities of audio files, as an advanced noise filtering step previous to automatic processing by recognition engines.

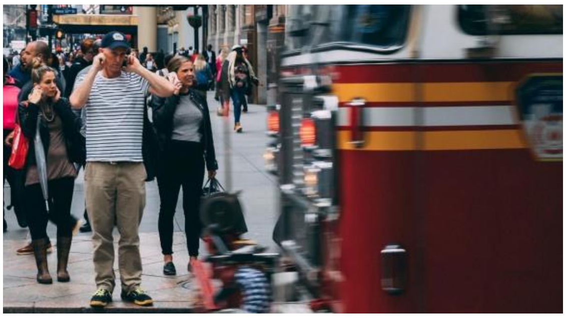
Because all model training is being conducted internally by our engineers, if required ***AudioFocus*** **can be customized** for maximizing its filtering performance against difficult to remove specific noises, or even for extracting a new type of audio signal not related to speech.

Two different features are included:

- **Denoise**: remove periodic and also spontaneous background noises, isolating the speech signal.
- **Enhance** (available soon): improve the perceived quality of the audio channel for better and more comfortable understanding by human operators, simulating different features: higher sampling rate, better frequency response and sensitivity of the microphone and a reverb free acoustic environment.

AudioFocus is distributed as a **SDK (*Software Development Kit*)** that exports its functionalities through a powerful yet easy to use **API** (*Application Programming Interface*). This highly efficient C++ API allows easy integration into any final application and operating environment.

AudioFocus can be an invaluable companion for maximizing results, enhancing audio in:



- **Real time communications in call centers**: reduce fatigue and improve understanding by human agents.
- **Online meetings**: remove background noises at home or outdoors.
- **Voice assistants and speaker authentication**: get higher accuracy rates in ASR and voice biometrics.
- **Large audio DB processing**: initial filtering previous to automatic processing.

PRODUCT

- Solution for removing static and dynamic noises from speech signals.

KEY FEATURES

- Can remove both **periodic** and **non-periodic** noises.
- **Can be customized** for dealing with especially problematic noises or extracting non-speech audio signals.
- Can run in **real time** or at file level for **batch operations**.
- **Easy to integrate API** for on-premise solutions.
- **Highly optimized C++ recognition engine**: can be integrated into embedded systems.

TECHNICAL SPECIFICATIONS

- Maximum audio length: unlimited (processing window can be configured).
- Processing speed: 1.5X real time (using the recommended CPU below).
- Disk space required for installation: 300 MB.
- RAM usage: 1.2 GB (typical, depending on window length configuration).
- Supported audio formats: WAV PCM linear 16 bits, MP3.
- Supported sampling frequencies: 8-48 KHz (44.1 KHz recommended).
- Proprietary C++ API.
- Minimum recommended CPU: Intel Core i5-4460@3.20 GHz or equivalent.

SUPPORTED PLATFORMS

- Windows® 10, 11.
- Linux, several distributions.

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