## CLAIMS

[Claim 1]

A method of noise abatement within a signal stream containing unwanted signal referred to as noise, comprising:

acquiring a digitized noise signal from an environment in which the signal stream is present;

dividing, by a digital processor circuit, the digitized noise signal into a set of different frequency ranges;

subdividing at least one frequency range in the set of frequency ranges into a plurality of noise segments;

computing, by a digital processor circuit, a frequency domain representation for the digitized noise signal;

individually shifting the frequency domain representation of one or more of the noise segments which comprise the digitized noise signal in time using the digital processor circuit to obtain time shifted components, where quantity of time shift is dependent on a selected frequency in the frequency domain representation and takes in account both an in-air propagation time of the noise signal and a system propagation time associated with the throughput speed of the digital processor circuit and any associated equipment, where the in-air propagation time is transit time for a signal to travel though air from an input microphone to a feedback microphone;

combining the time shifted components from each noise segment to form an anti-noise signal; and

outputting the anti-noise signal into the signal stream to abate the noise through destructive interference.

[Claim 2]

The method of claim 1 further comprising subdividing each frequency range in the set of frequency ranges such that the number of noise segments for a given frequency range varies between frequency ranges in the set of frequency ranges.

[Claim 3]

The method of claim 1 further comprising selectively applying a phase correction to one or more of the noise segments which comprise the digitized noise signal.

[Claim 4]

The method of claim 1,

wherein subdividing the at least one frequency range in the set of frequency ranges into the plurality of noise segments comprises subdividing each frequency range in the set of frequency ranges into frequency band segments of a selected width, wherein for each frequency band segment, shifting, by the digital processor circuit, a phase of that frequency band segment by an amount dependent on a selected frequency of the frequency band segment.

[Claim 5]

The method of claim 1 further comprising selectively applying an amplitude scaling to one or more of the noise segments which comprise the digitized noise signal.

[Claim 6]

The method of claim 1 further comprising one of the following:

amplitude scaling in one frequency range without amplitude scaling in another frequency range in the set of frequency ranges; or

applying phase shift and amplitude adjustment on the plurality of noise segments.

[Claim 7]

The method of claim 1 further comprising converting the anti-noise signal into an analog signal and outputting the anti-noise signal into the signal stream by mixing it with the noise within the signal stream.

[Claim 8]

The method of claim 7 further comprising mixing the anti-noise signal with the noise within the signal stream by using an amplified loudspeaker or other transducer disposed within the signal stream.

[Claim 9]

A system for noise abatement in an audio signal stream which contains a noise signal, comprising:

a microphone configured to acquire a noise signal from an environment in which the audio signal stream is present;

an analog-to-digital converter configured to receive the noise signal from the microphone and operate to convert the noise signal to a digitized noise signal;

a digital signal processor configured to receive the digitized noise signal from the analog-to-digital converter and process the digitized noise signal by

dividing, by a digital processor circuit, the digitized noise signal into a set of different frequency ranges,

subdividing at least one frequency range in the set of frequency ranges into a plurality of noise segments,

computing a frequency domain representation of the digitized noise signal in frequency domain,

individually shifting the frequency domain representation of one or more noise segments which comprise the digitized noise signal in time to obtain time

shifted components, where quantity of time shift is dependent on a selected frequency in the frequency domain representation and takes in account both an in-air propagation time of the noise signal and a system propagation time associated with the throughput speed of the digital processor circuit, where the in-air propagation time is transit time for a signal to travel through air from the microphone to a feedback microphone, and

combining the time shifted components from each noise segment to form an anti-noise signal; and

a digital-to-analog converter configured to receive the anti-noise signal from the digital signal processor and operate to convert the anti-noise signal to an analog anti-noise signal.

[Claim 10]

The system of claim 9, wherein the digital signal processor is further configured to subdivide each frequency range in the set of frequency ranges such that the number of noise segments for a given frequency range varies between frequency ranges in the set of frequency ranges.

[Claim 11]

The system of claim 9, wherein the digital signal processor is configured to selectively apply a phase correction to one or more of the noise segments which comprise the digitized noise signal.

[Claim 12]

The system of claim 9, wherein the digital signal processor is configured to selectively apply an amplitude scaling to one or more of the noise segments which comprise the digitized noise signal.

[Claim 13]

The system of claim 9 further comprising a mixer configured to receive the audio signal stream and the analog anti-noise signal and operate to combine the audio signal stream with the analog anti-noise signal.

[Claim 14]

A method for calibrating a system for noise abatement as claimed in any one of claims 9 to 13 embedded in an audio device, the method comprising:

disabling the system;

acquiring, by the audio device, a known signal from an environment surrounding the audio device;

determining an in-air propagation time of the known signal while the system is disabled, where the in-air propagation time is transit time for a signal to travel through air from an input microphone of the audio device to a speaker of the audio device;

enabling the system; and

determining a system propagation time of the known signal through the system while the system is enabled, where the system propagation time is throughput speed of a digital processor circuit comprising the system.

[Claim 15]

The method of claim 14 further comprising

computing a system offset time by subtracting the system propagation time from the in-air propagation time; and

storing the system offset time in a computer memory of the audio device, where the computer memory is accessible to the system.

[Claim 16]

The method of 14 further comprising outputting, by the system, an anti-noise signal into a signal stream output by the audio device, where the anti-noise signal is constructed using the system offset time.

[Claim 17]

The method of 14, wherein determining in-air propagation time of the known signal further comprises

determining a time the known signal is captured by the input microphone;

determining a time the known signal is captured by a feedback microphone disposed proximate to the speaker of the audio device; and

calculating the in-air propagation time by subtracting the time the known signal is captured by the input microphone from the time the known signal is captured by the feedback microphone.

[Claim 18]

The method of claim 14, wherein determining the system propagation time further comprises

determining a time the known signal is captured by the input microphone;

determining a time the known signal is captured by a feedback microphone disposed proximate to the speaker of the audio device; and

calculating the system propagation time by subtracting the time the known signal is captured by the input microphone from the time the known signal is captured by the feedback microphone.

[Claim 19]

The method of claim 18 further comprising altering the known signal after the known signal is acquired by the audio device but prior to inputting the known signal into the system.

[Claim 20]

The method of claim 16 further comprising

segmenting the known signal into a plurality of signal segments, where each signal segment in the plurality of signal segments correlates to a different frequency range;

for each signal segment, calculating a segment time delay, where the segment time delay causes a 180 degree phase shift at center frequency of the frequency range associated with a given signal segment;

for each signal segment, calculating an applied time delay as an absolute value of the segment time delay minus the system offset time; and

for each signal segment, storing the applied time delay in the computer memory of the audio device.