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PART 1 - ATTORNEY/APPLICANT COPY
page 1 of 1

LGE EXHIBIT NO. 1011
Amazon v. Jawbone
U.S. Patent 11,122,357
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CONFIRMATION NO. 2839

53186
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FILING RECEIPT

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Date Mailed: 05/12/2008

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Applicant(s)

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Power of Attorney: The patent practitioners associated with Customer Number 53186

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Title

Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System

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Attorney Docket No. ALPH.P035P2 Patent

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Application Data Sheet;
Provisional Patent Application;
Drawings; and
Electronic payment of filing fee.

United States Patent Application for Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System

Inventor: Greg Burnett

RELATED APPLICATION

This application is related to United States (US) Patent Application Number 60/953,444, filed August 1, 2007.

TECHNICAL FIELD

The disclosure herein relates generally to noise suppression. In particular, this disclosure relates to noise suppression systems, devices, and methods for use in acoustic applications.

INCORPORATION BY REFERENCE

Each patent, patent application, and/or publication mentioned in this specification is herein incorporated by reference in its entirety to the same extent as if each individual patent, patent application, and/or publication was specifically and individually indicated to be incorporated by reference.

Abstract

Embodiments of a noise suppression system are described. A dual omnidirectional microphone array (DOMA) implementation of a noise suppression system is presented. Two virtual microphones are constructed using two physical omnidirectional microphones. The virtual directional microphones are then used along with a VAD signal and an adaptive noise suppression algorithm to remove a significant amount of noise without adversely affecting the speech. The array herein is significantly different from conventional arrays in that the only null in the virtual microphones of an embodiment is for the user's speech, not the noise. The array and corresponding algorithm of an embodiment significantly increase the signal-to-noise ratio (SNR) of the user's speech signal and have been demonstrated to be robust with respect to variations in array-to-user's-mouth distance and orientation as well as ambient temperature, to name a few.

Brief Description of Figures

Figure 1: A two-microphone adaptive noise suppression system.

Figure 2: Array and speech source (S) configuration, under an embodiment. The microphones are separated by a distance approximately equal to $2d_0$, and the speech source is located a distance d_s away from the midpoint of the array at an angle θ. The system is axially symmetric so only d_s and θ need be specified.

Figure 3: Flow diagram for a first order gradient microphone using two omnidirectional elements O1 and O2, under an embodiment. The output of each is delayed, multiplied by a constant, and subtracted from the other to get the output.

Figure 4: Linear response of V2 to a 1 kHz speech source at a distance of 0.1 m, under an embodiment. Note the null at 0 degrees, where the speech is normally located.

Figure 5: Linear response of V2 to a 1 kHz noise source at a distance of 1.0 m, under an embodiment. Note that there is no null – all noise sources are detected.

Figure 6: Linear response of V1 to a 1 kHz speech source at a distance of 0.1 m, under an embodiment. There is no null and the response for speech is greater than that shown in Figure 4.

Figure 7: Linear response of V1 to a 1 kHz noise source at a distance of 1.0 m, under an embodiment. There is no null and the response is very similar to V2 shown in Figure 5.

Figure 8: Linear response of V1 to a speech source at a distance of 0.1 m for frequencies of 100, 500, 1000, 2000, 3000, and 4000 Hz, under an embodiment.

Figure 9: Comparison of frequency responses for speech for the array of an embodiment and for a conventional cardioid microphone.

Figure 10: Speech response for V1 (top, dashed) and V2 (bottom, solid) vs. B with d_s assumed to be 0.1 m, under an embodiment. Note that the spatial null in V2 is rather broad.

Figure 11: Ratio of V1 / V2 speech responses shown in Figure 10 versus B, under an embodiment. Note that the ratio is above 10 dB for all $0.8 < B < 1.1$. This means that the physical β of the system need not be exactly modeled for good performance.

Figure 12: Plot of B vs. actual d_s assuming that d_s = 10 cm and theta = 0, under an embodiment.

Figure 13: Plot of B vs. theta with d_s = 10 cm and assuming d_s = 10 cm, under an embodiment.

Figure 14: Amplitude (top) and phase (bottom) response of N(s) with B = 1 and D = -7.2 μsec, under an embodiment. The resulting phase difference clearly affects high frequencies more than low.

Figure 15: Amplitude (top) and phase (bottom) response of N(s) with B = 1.2 and D = -7.2 μsec, under an embodiment. Non-unity B affects the entire frequency range.

Figure 16: Amplitude (top) and phase (bottom) response of the effect on the speech cancellation in V2 due to a mistake in the location of

the speech source with q1 = 0 degrees and q2 = 30 degrees, under an embodiment. Note that the cancellation is still below -10 dB for frequencies below 6 kHz.

Figure 17: Amplitude (top) and phase (bottom) response of the effect on the speech cancellation in V2 due to a mistake in the location of the speech source with q1 = 0 degrees and q2 = 45 degrees, under an embodiment. Now the cancellation is below -10 dB only for frequencies below about 2.8 kHz and a reduction in performance is expected.

Figure 18: Experimental esults for a $2d_0$ = 19 mm array using a linear β of 0.83 on a Bruel and Kjaer Head and Torso Simulator (HATS) in very loud (~85 dBA) music/speech noise environment, under an embodiment. The noise has been reduced by about 25 dB and the speech hardly affected, with no noticeable distortion.

Definitions

Unless otherwise specified, the following terms have the corresponding meanings in addition to any meaning or understanding they may convey to one skilled in the art:

Bleedthrough: The undesired presence of noise during speech.
Denoising: Removing unwanted noise from Mic1
Devoicing: Removing/distorting the desired speech from Mic1
Directional microphone (DM): A physical directional microphone that is vented on both sides of the sensing diaphragm.
Mic1 (M1): General designation for an adaptive noise suppression system microphone that usually contains more speech than noise.
Mic2 (M2): General designation for an adaptive noise suppression system microphone that usually contains more noise than speech.
Noise: Unwanted environmental acoustic noise.
Null: A zero or minima in the spatial response of a physical or virtual directional microphone.
O1, O2: The physical omnidirectional microphones used to form the array
Speech: Desired speech of the user.
SSM: Skin Surface Microphone used in an earpiece (e.g., the Jawbone earpiece available from Aliph of San Francisco, California) to detect speech vibrations on the user's skin.
V1: The virtual directional "speech" microphone – has no nulls.
V2: The virtual directional "noise" microphone – has a null for the user's speech.
VAD: Voice Activity Detection signal - indicates when user speech is detected.
Virtual directional microphones (VM): A microphone constructed using two or more omnidirectional microphones and signal processing.

I. Background

Adaptive noise suppression algorithms have been around for some time. The basic algorithmic flow is shown in **Figure 1**. Two or more microphones are used to sample both an (unwanted) acoustic noise field and the (desired) speech of a user. The noise relationship between the microphones is then determined using an adaptive filter (such as Least-Mean-Squares as described in Haykin & Widrow, ISBN# 0471215708, Wiley, 2002, but any adaptive or stationary system identification algorithm may be used) and that relationship used to filter the noise from the desired signal.

Most of the noise suppression systems in use today for speech communication systems are based on a single-microphone spectral subtraction technique first develop in the 1970's and described, for example, by S. F. Boll in "Suppression of Acoustic Noise in Speech using Spectral Subtraction," IEEE Trans. on ASSP, pp. 113-120, 1979. These techniques have been refined over the years, but the basic principles of operation have remained the same. See, for example, United States Patent Number 5,687,243 of McLaughlin, et al., and United States Patent Number 4,811,404 of Vilmur, et al. There have also been several attempts at multi-microphone noise suppression systems, such as those outlined in United States Patent Number 5,406,622 of Silverberg et al. and United States Patent Number 5,463,694 of Bradley et al. Multi-microphone systems have not been very successful for a variety of reasons, the most compelling being poor noise cancellation performance and/or significant speech distortion. Primarily, conventional multi-microphone systems attempt to increases the SNR of the user's speech by "steering" the nulls of the system to the strongest noise sources. This approach is limited in the number of noise sources removed by the number of available nulls.

The Jawbone earpiece, introduced by Aliphcom in December 2006, was the first known commercial product to use a pair of physical directional microphones (instead of omnidirectional microphones) to reduce environmental acoustic noise.

The technology behind the Jawbone is currently described under one or more of US Patent Number 7,246,058 by Burnett and/or US Patent Application Numbers 10/400,282, 10/667,207, and/or 10/769,302. Generally, multi-microphone techniques make use of an acoustic-based Voice Activity Detector (VAD) to determine the background noise characteristics, where "voice" is generally understood to include human voiced speech, unvoiced speech, or a combination of voiced and unvoiced speech. The Jawbone improved on this by using a microphone-based sensor to construct a VAD signal using directly detected speech vibrations in the user's cheek. This allowed the Jawbone to aggressively remove noise when the user was not producing speech. However, the current Jawbone product uses a directional microphone array which can be difficult to manufacture reliably, and an omnidirectional array replacement was desired.

II. Adaptive noise suppression basics

In analyzing the single noise source 101 and the direct path to the microphones, with reference to **Figure 1**, the total acoustic information coming into MIC 1 (102, which can be an physical or virtual microphone) is denoted by $m_1(n)$. The total acoustic information coming into MIC 2 (103, which can also be an physical or virtual microphone) is similarly labeled $m_2(n)$. In the z (digital frequency) domain, these are represented as $M_1(z)$ and $M_2(z)$. Then,

$$M_1(z) = S(z) + N_2(z)$$
$$M_2(z) = N(z) + S_2(z)$$

with

$$N_2(z) = N(z)H_1(z)$$
$$S_2(z) = S(z)H_2(z),$$

so that

$$M_1(z) = S(z) + N(z)H_1(z)$$
$$M_2(z) = N(z) + S(z)H_2(z). \qquad \text{Eq. 1}$$

This is the general case for all two microphone systems. In a practical system there is always going to be some leakage of noise into MIC 1, and some leakage of

speech into MIC 2. Equation 1 has four unknowns and only two known relationships and therefore cannot be solved explicitly.

However, there is another way to solve for some of the unknowns in Equation 1. The analysis starts with an examination of the case where the speech is not being generated, that is, where a signal from the VAD subsystem 104 equals zero. In this case, s(n) = S(z) = 0, and Equation 1 reduces to

$$M_{1N}(z)=N(z)H_1(z)$$
$$M_{2N}(z)=N(z),$$

where the N subscript on the M variables indicate that only noise is being received. This leads to

$$M_{1N}(z)=M_{2N}(z)H_1(z)$$
$$H_1(z)=\frac{M_{1N}(z)}{M_{2N}(z)}. \qquad \text{Eq. 2}$$

The function $H_1(z)$ can be calculated using any of the available system identification algorithms and the microphone outputs when the system is certain that only noise is being received. The calculation can be done adaptively, so that the system can react to changes in the noise.

A solution is now available for $H_1(z)$, one of the unknowns in Equation 1. The final unknown, $H_2(z)$, can be determined by using the instances where speech is being produced and the VAD equals one. When this is occurring, but the recent (perhaps less than 1 second) history of the microphones indicate low levels of noise, it can be assumed that n(s) = N(z) ~ 0. Then Equation 1 reduces to

$$M_{1S}(z)=S(z)$$
$$M_{2S}(z)=S(z)H_2(z),$$

which in turn leads to

$$M_{2S}(z)=M_{1S}(z)H_2(z)$$

$$H_2(z)=\frac{M_{2S}(z)}{M_{1S}(z)},$$

which is the inverse of the $H_1(z)$ calculation. However, it is noted that different inputs are being used (now only the speech is occurring whereas before only the noise was occurring). While calculating $H_2(z)$, the values calculated for $H_1(z)$ are held constant (and vice versa) and it is assumed that the noise level is not high enough to cause errors in the $H_2(z)$ calculation.

After calculating $H_1(z)$ and $H_2(z)$, they are used to remove the noise from the signal. If Equation 1 is rewritten as

$$\begin{aligned} S(z)&=M_1(z)-N(z)H_1(z) \\ N(z)&=M_2(z)-S(z)H_2(z) \\ S(z)&=M_1(z)-[M_2(z)-S(z)H_2(z)]H_1(z) \\ S(z)[1-H_2(z)H_1(z)]&=M_1(z)-M_2(z)H_1(z), \end{aligned}$$

then N(z) may be substituted as shown to solve for S(z) as

$$S(z)=\frac{M_1(z)-M_2(z)H_1(z)}{1-H_1(z)H_2(z)}. \qquad \text{Eq. 3}$$

If the transfer functions $H_1(z)$ and $H_2(z)$ can be described with sufficient accuracy, then the noise can be completely removed and the original signal recovered. This remains true without respect to the amplitude or spectral characteristics of the noise. If there is very little or no leakage from the speech source into M_2, then $H_2(z)\approx 0$ and Equation 3 reduces to

$$S(z)\approx M_1(z)-M_2(z)H_1(z). \qquad \text{Eq. 4}$$

This is much simpler to implement and is very stable, assuming $H_1(z)$ is stable. However, if significant speech energy is in $M_2(z)$, devoicing can occur. In order to construct a well-performing system and use Equation 4, consideration is given to the following:

R1. Availability of a perfect (or at least very good) VAD in noisy conditions

R2. Sufficiently accurate $H_1(z)$

R3. Very small (ideally zero) $H_2(z)$.

R4. During speech production, $H_1(z)$ cannot change substantially.

R5. During noise, $H_2(z)$ cannot change substantially.

R1 is easy to satisfy if the SNR of the desired speech to the unwanted noise is high enough. "Enough" means different things depending on the method of VAD generation. If a VAD vibration sensor is used, as in Burnett 7,256,048, accurate VAD in very low SNRs (-10 dB or less) is possible. Acoustic–only methods using information from O_1 and O_2 can also return accurate VADs, but are limited to SNRs of ~3 dB or greater for adequate performance.

R5 is normally simple to satisfy because for most applications the microphones will not change position with respect to the user's mouth very often or rapidly. In those applications where it may happen (such as hands-free conferencing systems) it can be satisfied by configuring Mic2 so that $H_2(z) \approx 0$.

Satisfying R2, R3, and R4 are more difficult but are possible given the right combination of V_1 and V_2. Methods are examined below that have proven to be effective in satisfying the above, resulting in excellent noise suppression performance and minimal speech removal and distortion in an embodiment.

III. A different omnidirectional array solution

A generalized two-microphone array is shown below in **Figure 2**. Two omnidirectional microphones 201 and 202 are placed a distance $2d_0$ apart and a speech source 200 is located a distance d_s away at an angle of θ. This array is axially symmetric (at least in free space), so no other angle is needed. The output from each microphone can be delayed (z_1 and z_2), multiplied by a gain (A_1 and A_2), and then summed with the other as demonstrated in **Figure 3**. This operation can

be over any frequency range desired. By varying the magnitude and sign of the delays and gains, a wide variety of virtual directional microphones (VMs) can be realized. There are other methods known to those skilled in the art for constructing VMs but this is a common one and will be used in the enablement below.

The construction of VMs for the adaptive noise suppression system of an embodiment includes:

1. Very similar noise response in V1 and V2. This means that $H_1(z)$ is simple to model and will not change much during speech, satisfying R2 and R4 above and allowing strong denoising and minimized bleedthrough.
2. Very little speech response for V2. This means that $H_2(z) \approx 0$, which will satisfy R3 and R5 above.
3. Sufficient speech response for V1 so that the cleaned speech will have significantly higher SNR than the original speech captured by O1.

It is assumed below that the responses of the omnidirectional microphones O1 and O2 to an identical acoustic source have been normalized so that they have exactly the same response (amplitude and phase) to that source. This can be accomplished using standard microphone array methods (such as frequency-based calibration) well known to those versed in the art.

Using the second guideline of the VMs as a starting point (e.g., very little speech response for V2), it is seen that for discrete systems $V_2(z)$ can be represented as:

$$V_2(z) = O_2(z) - z^{-\gamma}\beta O_1(z)$$

where

$$\beta = \frac{d_1}{d_2}$$

$$\gamma = \frac{d_2 - d_1}{c} \cdot f_s \quad \text{(samples)}$$

$$d_1 = \sqrt{d_s^2 - 2d_s d_0 \cos(\theta) + d_0^2}$$
$$d_2 = \sqrt{d_s^2 + 2d_s d_0 \cos(\theta) + d_0^2}$$

The distances d_1 and d_2 are the distance from O_1 and O_2 to the speech source, respectively, and γ is their difference divided by *c*, the speed of sound, and multiplied by the sampling frequency f_s. Thus γ is in samples, but need not be an integer. For non-integer γ, fractional-delay filters (well known to those versed in the art) may be used.

It is important to note that the β above is NOT the conventional β used to denote the mixing of VMs in adaptive beamforming; it is a physical variable of the system that depends on the intra-microphone distance d_0 (which is fixed) and the distance d_s and angle θ, which can vary. As shown below, for properly calibrated microphones, it is not necessary for the system to be programmed with the exact β of the array. Errors of approximately 10-15% in the actual β (i.e. the β used by the algorithm is not the β of the physical array) have been used with very little degradation in quality. The algorithmic value of β may be calculated and set for a particular user or may be calculated adaptively during speech production when little or no noise is present. However, adaptation during use is not required for nominal performance.

The above formulation for $V_2(z)$ has a null at the speech location and will therefore exhibit minimal response to the speech. This is shown in **Figure 4** for an array with d_0 = 10.7 mm and a speech source on the axis of the array (θ = 0) at 10 cm. Note that the speech null at zero degrees is not present for noise in the far field, as shown in **Figure 5** with a noise source distance of approximately 1 meter. This insures that noise in front of the user will be detected so that it can be removed. This differs from conventional systems that can have difficulty removing noise in the direction of the mouth of the user.

The $V_1(z)$ can be formulated using the general form for $V_1(z)$:

$$V_1(z) = \alpha_A O_1(z) \cdot z^{-d_A} - \alpha_B O_2(z) \cdot z^{-d_B}$$

Since

$$V_2(z) = O_2(z) - z^{-\gamma} \beta O_1(z)$$

then for noise in the forward direction

$$O_{2N}(z) = O_{1N}(z) \cdot z^{-\gamma}$$

and

$$V_{2N}(z) = O_{1N}(z) \cdot z^{-\gamma} - z^{-\gamma} \beta O_{1N}(z)$$
$$V_{2N}(z) = (1-\beta)\left(O_{1N}(z) \cdot z^{-\gamma}\right)$$

if this is set equal to $V_1(z)$ above, the result is

$$V_{1N}(z) = \alpha_A O_{1N}(z) \cdot z^{-d_A} - \alpha_B O_{2N}(z) \cdot z^{-d_B} = (1-\beta)\left(O_{1N}(z) \cdot z^{-\gamma}\right)$$

thus

$$d_A = \gamma$$
$$d_B = 0$$
$$\alpha_A = 1$$
$$\alpha_B = \beta$$

and

$$V_1(z) = O_1(z) \cdot z^{-\gamma} - \beta O_2(z)$$

This formulation assures that the noise response will be as similar as possible and that the speech response will be proportional to $(1-\beta^2)$. Since β is the ratio of the distances from O1 and O2 to the speech source, it is affected by the size of the array and the distance from the array to the speech source.

The response of V_1 to speech is shown in **Figure 6**, and the response to noise in **Figure 7**. Note the difference in speech response compared to V_2 shown in **Figure 4** and the similarity of noise response shown in **Figure 5**. Also note that the orientation of the speech response for V_1 shown in **Figure 6** is completely opposite the orientation of conventional systems, where the main lobe of response is normally oriented toward the speech source. The orientation of an embodiment, in which the main lobe of the speech response of V_1 is oriented away from the speech source, means that the speech sensitivity of V_1 is lower than a normal directional microphone but is flat for all frequencies within approximately +-30 degrees of the axis of the array, as shown in **Figure 8**. This flatness of response for speech means that no shaping postfilter is needed to restore omnidirectional frequency response. This does come at a price - as shown in **Figure 9**, which shows the speech response of V_1 and the speech response of a cardioid microphone. The speech response of V_1 is approximately 0 to ~13 dB less than a normal directional microphone between approximately 500 and 7500 Hz and approximately 0 to 10+ dB greater than a directional microphone below approximately 500 Hz and above 7500 Hz for a sampling frequency of approximately 16000 Hz. However, the superior noise suppression made possible using this system more than compensates for the initially poorer SNR.

It should be noted that **Figures 4-7** assume the speech is located at approximately 0 degrees and approximately 10 cm and the noise at all angles and approximately 1.0 meters. Generally, the noise distance is not required to be 1 m or more, but the denoising is the best for those distances. For distances less than approximately 1 m, denoising will not be as effective due to the greater dissimilarity in the noise responses of V1 and V2. This has not proven to be an impediment in practical use – in fact, it can be seen as a feature. Any "noise" source that is ~10 cm away from the earpiece is likely to be desired to be captured and transmitted.

The definitions for V_1 and V_2 above mean that for noise $H_1(z)$ is:

$$H_1(z) = \frac{V_1(z)}{V_2(z)} = \frac{-\beta O_2(z) + O_1(z) \cdot z^{-\gamma}}{O_2(z) - z^{-\gamma}\beta O_1(z)}$$

which has the form of an allpass filter, which has the advantage of being easily and accurately modeled, especially in magnitude response, satisfying R2.

Clearly the speech null designed into V_2 means that the VAD signal is no longer a critical component. The VAD's purpose was to ensure that the system would not train on speech and then subsequently remove it, resulting in speech distortion. If, however, V_2 contains no speech, the adaptive system cannot train on the speech and cannot remove it. As a result, the system can denoise all the time without fear of devoicing, and the resulting clean audio can then be used to generate a VAD signal for use in subsequent single-channel noise suppression algorithms such as spectral subtraction. In addition, constraints on the absolute value of $H_1(z)$ (i.e. restricting it to absolute values less than two) can keep the system from fully training on speech even if it is detected. In reality, though, speech can be present due to a mis-located V_2 null and/or echoes or other phenomena, and a VAD sensor or other acoustic-only VAD is recommended to minimize speech distortion.

Errors in estimating β and γ

Depending on the application, β and γ may be fixed in the noise suppression algorithm or they can be estimated when the algorithm indicates that speech production is taking place in the presence of little or no noise. In either case, there is likely to be an error in the estimate of the actual β and γ of the system. This section examines these errors and their effect on the performance of the system. As above, "good performance" of the system indicates that there is sufficient denoising and minimal devoicing.

The effect of an incorrect β and γ on the response of V1 and V2 can be seen by examining the definitions above:

$$V_1(z) = O_1(z) \cdot z^{-\gamma_T} - \beta_T O_2(z)$$
$$V_2(z) = O_2(z) - z^{-\gamma_T} \beta_T O_1(z)$$

where β_T and γ_T denote the theoretical estimates of β and γ used in the noise suppression algorithm. In reality, the speech response of O2 is

$$O_{2S}(z) = \beta_R O_{1S}(z) \cdot z^{-\gamma_R}$$

where β_R and γ_R denote the real β and γ of the physical system. The differences between the theoretical and actual values of β and γ can be due to mis-location of the speech source (it is not where it is assumed to be) and/or a change in air temperature (which changes the speed of sound). Inserting the actual response of O2 for speech into the above equations for V1 and V2 yields

$$V_{1S}(z) = O_{1S}(z)\left[z^{-\gamma_T} - \beta_T \beta_R z^{-\gamma_R}\right]$$
$$V_{2S}(z) = O_{1S}(z)\left[\beta_R z^{-\gamma_R} - \beta_T z^{-\gamma_T}\right]$$

If the difference in phase is represented by

$$\gamma_R = \gamma_T + \gamma_D$$

And the difference in amplitude as

$$\beta_R = B\beta_T$$

then

$$\begin{aligned} V_{1S}(z) &= O_{1S}(z) z^{-\gamma_T}\left[1 - B\beta_T^2 z^{-\gamma_D}\right] \\ V_{2S}(z) &= \beta_T O_{1S}(z) z^{-\gamma_T}\left[B z^{-\gamma_D} - 1\right] \end{aligned} \quad \text{Eq. 5}$$

Clearly the speech cancellation in V2 (which directly affects the degree of devoicing) and the speech response of V1 will be dependent on both B and D. An examination of the case where D = 0 follows. In **Figure 10**, the speech response for V1 (upper, dashed) and V2 (lower, solid) compared to O1 is shown versus B when d_s is thought to be approximately 10 cm and $\theta = 0$. When B = 1, the speech

is absent from V2. In **Figure 11**, the ratio of the speech responses in Figure 10 is shown. When 0.8 < B < 1.1, the V1/V2 ratio is above approximately 10 dB – enough for good performance. Clearly, if D = 0, B can vary significantly without adversely affecting the performance of the system. Again, this assumes that calibration of the microphones so that both their amplitude and phase response is the same for an identical source has been performed.

The B factor can be non-unity for a variety of reasons. Either the distance to the speech source or the relative orientation of the array axis and the speech source or both can be different than expected. If both distance and angle mismatches are included for B, then

$$B = \frac{\beta_R}{\beta_T} \frac{\sqrt{d_{SR}^2 - 2d_{SR}d_0\cos(\theta_R) + d_0^2}}{\sqrt{d_{SR}^2 + 2d_{SR}d_0\cos(\theta_R) + d_0^2}} \cdot \frac{\sqrt{d_{ST}^2 + 2d_{ST}d_0\cos(\theta_T) + d_0^2}}{\sqrt{d_{ST}^2 - 2d_{ST}d_0\cos(\theta_T) + d_0^2}}$$

where again the T subscripts indicate the theorized values and R the actual values. In **Figure 12**, the factor B is plotted with respect to the actual d_s with the assumption that d_s = 10 cm and θ = 0. So, if the speech source in on-axis of the array, the actual distance can vary from approximately 5 cm to 18 cm without significantly affecting performance – a significant amount. Similarly, **Figure 13** shows what happens if the speech source is located at a distance of approximately 10 cm but not on the axis of the array. In this case, the angle can vary up to approximately +-55 degrees and still result in a B less than 1.1, assuring good performance. This is a significant amount of allowable angular deviation. If there is both angular and distance errors, the equation above may be used to determine if the deviations will result in adequate performance. Of course, if the value for β_T is allowed to update during speech, essentially tracking the speech source, then B can be kept near unity for almost all configurations.

An examination follows of the case where B is unity but D is nonzero. This can happen if the speech source is not where it is thought to be or if the speed of sound

is different from what it is believed to be. From the Equation 5 above, it can be sees that the factor that weakens the speech null in V2 for speech is

$$N(z) = Bz^{-\gamma D} - 1$$

or in the continuous s domain

$$N(s) = Be^{-Ds} - 1.$$

Since γ is the time difference between arrival of speech at V1 compared to V2, it can be errors in estimation of the angular location of the speech source with respect to the axis of the array and/or by temperature changes. Examining the temperature sensitivity, the speed of sound varies with temperature as

$$c = 331.3 + (0.606T) \text{ m/s}$$

where T is degrees Celsius. As the temperature decreases, the speed of sound also decreases. Setting 20 C as a design temperature and a maximum expected temperature range to -40 C to +60 C (-40 F to 140 F). The design speed of sound at 20 C is 343 m/s and the slowest speed of sound will be 307 m/s at -40 C with the fastest speed of sound 362 m/s at 60 C. Set the array length ($2d_0$) to be 21 mm. For speech sources on the axis of the array, the difference in travel time for the largest change in the speed of sound is

$$\nabla t_{MAX} = \frac{d}{c_1} - \frac{d}{c_2} = 0.021\,\text{m}\left(\frac{1}{343\,\text{m/s}} - \frac{1}{307\,\text{m/s}}\right) = -7.2\,\text{x}\,10^{-6}\text{ sec}$$

or approximately 7 microseconds. The response for N(s) given B = 1 and D = 7.2 μsec is shown in **Figure 14**. The amplitude response is less than approximately -10 dB for all frequencies less than 7 kHz and is only about -9 dB at 8 kHz. Therefore, assuming B = 1, this system would likely perform well at frequencies up to approximately 8 kHz. This means that a properly compensated system would work well even up to 8 kHz in an exceptionally wide (e.g., -40 C to 80 C)

temperature range. Note that the phase mismatch due to the delay estimation error causes N(s) to be much larger at high frequencies compared to low.

If B is not unity, the robustness of the system is reduced since the effect from non-unity B is cumulative with that of non-zero D. **Figure 15** shows the amplitude and phase response for B = 1.2 and D = 7.2 μsec. Now N(s) is below approximately -10 dB only for frequencies less than approximately 5 kHz and the response at low frequencies is much larger. Such a system would still perform well below 5 kHz and would only suffer from slightly elevated devoicing for frequencies above 5 kHz. For ultimate performance, a temperature sensor may be integrated into the system to allow the algorithm to adjust γ_T as the temperature varies.

Another way in which D can be non-zero is when the speech source is not where it is believed to be – specifically, the angle from the axis of the array to the speech source is incorrect. The distance to the source may be incorrect as well, but that introduces an error in B, not D.

Referring to **Figure 2**, it can be seen that for two speech sources (each with their own d_s and θ) that the time difference between the arrival of the speech at O1 and the arrival at O2 is

$$\Delta t = \frac{1}{c}\left(d_{12} - d_{11} - d_{22} + d_{21}\right)$$

where

$$d_{11} = \sqrt{d_{S1}^2 - 2d_{S1}d_0\cos(\theta_1) + d_0^2}$$
$$d_{12} = \sqrt{d_{S1}^2 + 2d_{S1}d_0\cos(\theta_1) + d_0^2}$$
$$d_{21} = \sqrt{d_{S2}^2 - 2d_{S2}d_0\cos(\theta_2) + d_0^2}$$
$$d_{22} = \sqrt{d_{S2}^2 + 2d_{S2}d_0\cos(\theta_2) + d_0^2}$$

The V2 speech cancellation response for θ_1 = 0 degrees and θ_2 = 30 degrees and assuming that B = 1 is shown in **Figure 16**. The cancellation is still below approximately -10 dB for frequencies below approximately 6 kHz, so an error of this

type will not significantly affect the performance of the system. However, if θ_2 is increased to approximately 45 degrees, as shown in **Figure 17**, the cancellation is below approximately -10 dB only for frequencies below approximately 2.8 kHz. The poor V2 speech cancellation above approximately 4 kHz may result in significant devoicing for those frequencies.

Alternate microphone calibration technique

The above has assumed that the microphones O1 and O2 were calibrated so that their response to a source located the same distance away was identical for both amplitude and phase. This is not always feasible, so a more practical calibration procedure is presented below. It is not as accurate, but is much simpler to implement. Begin by defining a filter $\alpha(z)$ such that:

$$O_{1C}(z) = \alpha(z) O_{2C}(z)$$

where the "C" subscript indicates the use of a known calibration source. The simplest one to use is the speech of the user. Then

$$O_{1S}(z) = \alpha(z) O_{2C}(z)$$

The microphone definitions are now:

$$V_1(z) = O_1(z) \cdot z^{-\gamma} - \beta(z)\alpha(z)O_2(z)$$
$$V_2(z) = \alpha(z)O_2(z) - z^{-\gamma}\beta(z)O_1(z)$$

The β of the system should be fixed and as close to the real value as possible. In practice, the system is not sensitive to changes in β and errors of approximately +-5% are easily tolerated. During times when the user is producing speech but there is little or no noise, the system can train $\alpha(z)$ to remove as much speech as possible. This is accomplished by:

1. Construct an adaptive system as shown in **Figure 1** with $\beta O_{1S}(z)z^{-\gamma}$ in the "MIC1" position, $O_{2S}(z)$ in the "MIC2" position, and $\alpha(z)$ in the $H_1(z)$ position.
2. During speech, adapt $\alpha(z)$ to minimize the residual of the system.
3. Construct $V_1(z)$ and $V_2(z)$ as above.

A simple adaptive filter can be used for $\alpha(z)$ so that only the relationship between the microphones is well modeled. The system of an embodiment trains only when speech is being produced by the user. A sensor like the SSM is invaluable in determining when speech is being produced in the absence of noise. If the speech source is fixed in position and will not vary significantly during use (such as when the array is on an earpiece), the adaptation should be infrequent and slow to update in order to minimize any errors introduced by noise present during training.

Variations in virtual microphone patterns

The above formulation works very well because the noise (far-field) responses of V1 and V2 are very similar while the speech (near-field) responses are very different. However, the formulations for V1 and V2 can be varied and still result in good performance of the system as a whole. If the definitions for V1 and V2 are taken from above and new variables B1 and B2 are inserted, the result is:

$$V_1(z) = O_1(z) \cdot z^{-\gamma_I} - B_1 \beta_I O_2(z)$$
$$V_2(z) = O_2(z) - z^{-\gamma_I} B_2 \beta_I O_1(z)$$

where B1 and B2 are both positive numbers or zero. If B1 and B2 are set equal to unity, the optimal system results as described above. If B1 is allowed to vary from unity, the response of V1 is affected. An examination of the case where B2 is left at 1 and B1 is decreased follows. As B1 drops to approximately zero, V1 becomes less and less directional, until it becomes a simple omnidirectional microphone when B1 = 0. Since B2 = 1, a speech null remains in V2, so very different speech responses remain for V1 and V2. However, the noise responses are much less similar, so denoising will not be as effective. Practically, though, the system still performs well. B1 can also be increased from unity and once again the system will still denoise well, just not as well as with B1 = 1.

If B2 is allowed to vary, the speech null in V2 is affected. As long as the speech null is still sufficiently deep, the system will still perform well. Practically values down to approximately B2 = 0.6 have shown sufficient performance, but it is recommended to set B2 close to unity for optimal performance.

In conclusion, the system is flexible enough to operate well at a variety of B1 values, but B2 values should be close to unity to limit devoicing for best performance.

Experimental results

Results for a $2d_0$ = 19 mm array using a linear β of 0.83 and B1 = B2 = 1 on a Bruel and Kjaer Head and Torso Simulator (HATS) in very loud (~85 dBA) music/speech noise environment are shown in **Figure 18**. The alternate microphone calibration technique discussed above was used to calibrate the microphones. The noise has been reduced by about 25 dB and the speech hardly affected, with no noticeable distortion. Clearly the technique significantly increases the SNR of the original speech, far outperforming conventional noise suppression techniques.

The DOMA of an embodiment can include algorithms or programs that run under and/or in association with a processing system. The processing system includes any collection of processor-based devices or computing devices operating together, or components of processing systems or devices, as is known in the art. For example, the processing system can include one or more of a processor, processor-based device, portable computer, portable communication device operating in a communication network, and/or a network server. The portable computer can be any of a number and/or combination of devices selected from among personal computers, cellular telephones, personal digital assistants, portable computing devices, and portable communication devices, but is not so limited. The processing system can include components within a larger computer system.

The processing system of an embodiment includes at least one processor. The processing system can also include or be coupled to at least one database. The term "processor" as generally used herein refers to any logic processing unit, such as one or more central processing units (CPUs), digital signal processors (DSPs), application-specific integrated circuits (ASIC), etc. The processor and memory can be monolithically integrated onto a single chip, distributed among a number of chips or components of the DOMA, and/or provided by some combination of algorithms. The DOMA methods described herein can be implemented in one or more of

software algorithm(s), programs, firmware, hardware, components, circuitry, in any combination.

Aspects of the DOMA described herein may be implemented as functionality programmed into any of a variety of circuitry, including programmable logic devices (PLDs), such as field programmable gate arrays (FPGAs), programmable array logic (PAL) devices, electrically programmable logic and memory devices and standard cell-based devices, as well as application specific integrated circuits (ASICs). Some other possibilities for implementing aspects of the DOMA include: microcontrollers with memory (such as electronically erasable programmable read only memory (EEPROM)), embedded microprocessors, firmware, software, etc. Furthermore, aspects of the DOMA may be embodied in microprocessors having software-based circuit emulation, discrete logic (sequential and combinatorial), custom devices, fuzzy (neural) logic, quantum devices, and hybrids of any of the above device types. Of course the underlying device technologies may be provided in a variety of component types, e.g., metal-oxide semiconductor field-effect transistor (MOSFET) technologies like complementary metal-oxide semiconductor (CMOS), bipolar technologies like emitter-coupled logic (ECL), polymer technologies (e.g., silicon-conjugated polymer and metal-conjugated polymer-metal structures), mixed analog and digital, etc.

It should be noted that any system, method, and/or other components disclosed herein may be described using computer aided design tools and expressed (or represented), as data and/or instructions embodied in various computer-readable media, in terms of their behavioral, register transfer, logic component, transistor, layout geometries, and/or other characteristics. Computer-readable media in which such formatted data and/or instructions may be embodied include, but are not limited to, non-volatile storage media in various forms (e.g., optical, magnetic or semiconductor storage media) and carrier waves that may be used to transfer such formatted data and/or instructions through wireless, optical, or wired signaling media or any combination thereof. Examples of transfers of such formatted data and/or instructions by carrier waves include, but are not limited to, transfers

(uploads, downloads, e-mail, etc.) over the Internet and/or other computer networks via one or more data transfer protocols (e.g., HTTP, FTP, SMTP, etc.). When received within a computer system via one or more computer-readable media, such data and/or instruction-based expressions of the above described components may be processed by a processing entity (e.g., one or more processors) within the computer system in conjunction with execution of one or more other computer programs.

Unless the context clearly requires otherwise, throughout the description, the words "comprise," "comprising," and the like are to be construed in an inclusive sense as opposed to an exclusive or exhaustive sense; that is to say, in a sense of "including, but not limited to." Words using the singular or plural number also include the plural or singular number respectively. Additionally, the words "herein," "hereunder," "above," "below," and words of similar import, when used in this application, refer to this application as a whole and not to any particular portions of this application. When the word "or" is used in reference to a list of two or more items, that word covers all of the following interpretations of the word: any of the items in the list, all of the items in the list and any combination of the items in the list.

The above description of embodiments of the DOMA is not intended to be exhaustive or to limit the systems and methods to the precise forms disclosed. While specific embodiments of, and examples for, the DOMA are described herein for illustrative purposes, various equivalent modifications are possible within the scope of the systems and methods, as those skilled in the relevant art will recognize. The teachings of the DOMA provided herein can be applied to other systems and methods, not only for the systems and methods described above. The elements and acts of the various embodiments described above can be combined to provide further embodiments. These and other changes can be made to the DOMA in light of the above detailed description.

Conclusion

A dual omnidirectional microphone array noise suppression technique has been described. Compared to conventional arrays and algorithms, which seek to reduce noise by nulling out noise sources, the array of an embodiment is used to form two distinct virtual directional microphones which are configured to have very similar noise responses and very dissimilar speech responses. The only null formed is one used to remove the speech of the user from V2. The two virtual microphones are paired with an adaptive filter algorithm and VAD algorithm to significantly reduce the noise without distorting the speech, significantly improving the SNR of the desired speech over conventional noise suppression systems. The embodiments described herein are simple to implement, stable in operation, flexible with respect to virtual microphone pattern choice, and have proven to be robust with respect to speech source-to-array distance and orientation as well as temperature and calibration techniques.

Figures

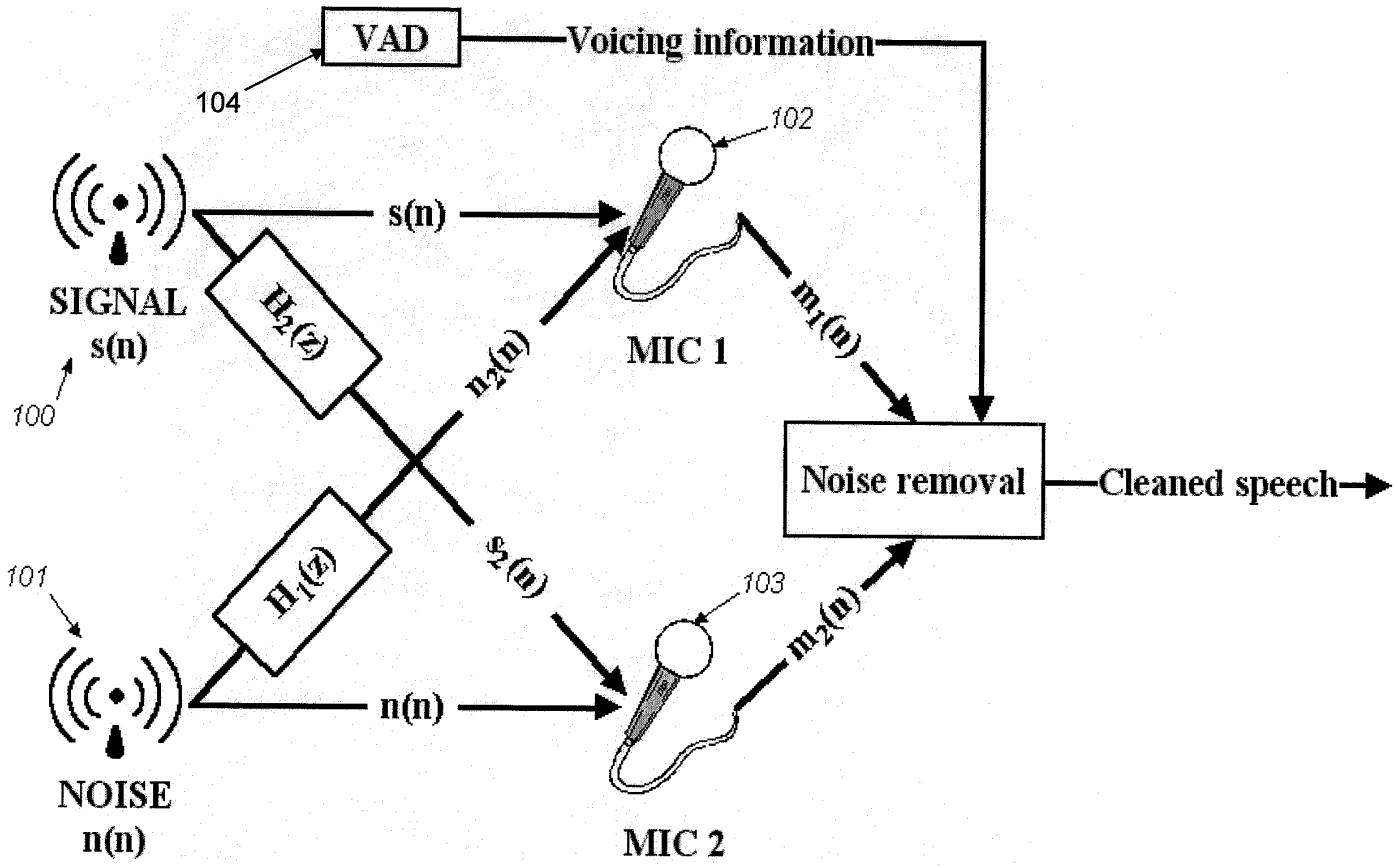

FIGURE 1

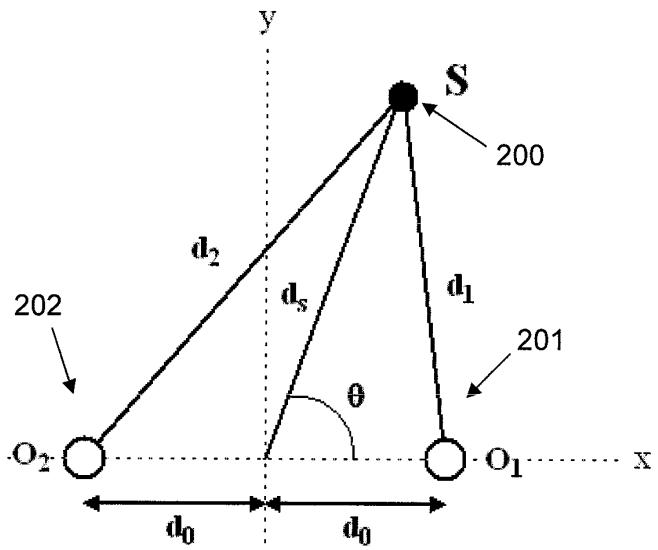

FIGURE 2

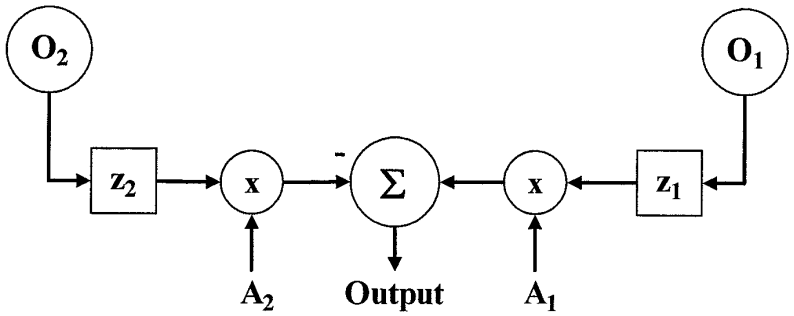

FIGURE 3

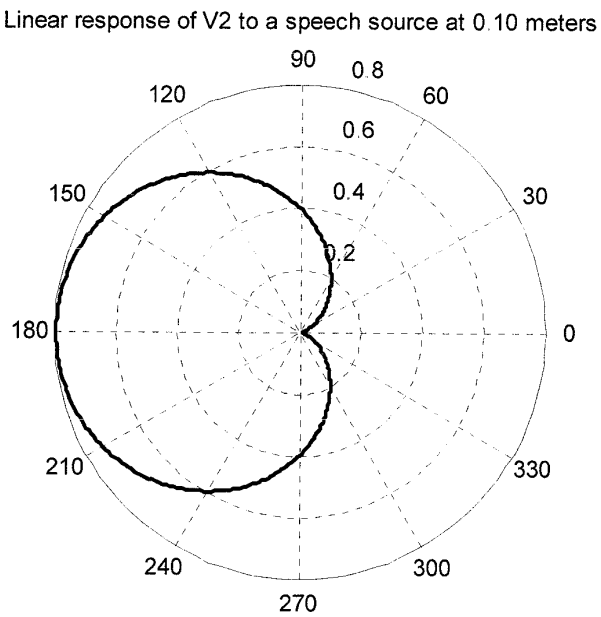

Figure 4

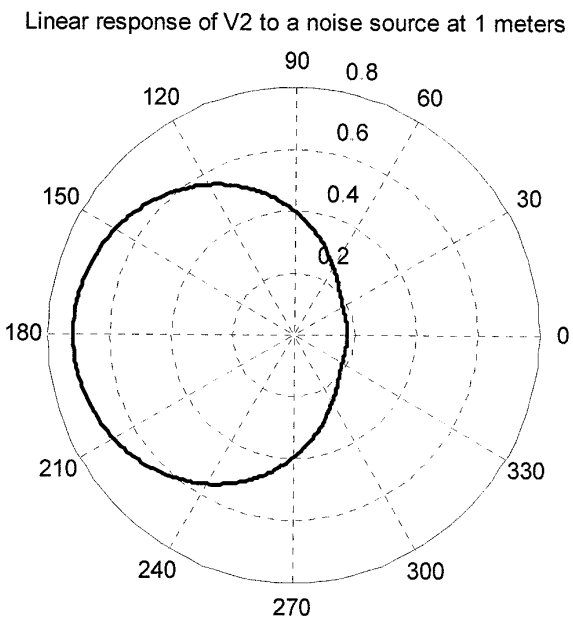

Figure 5

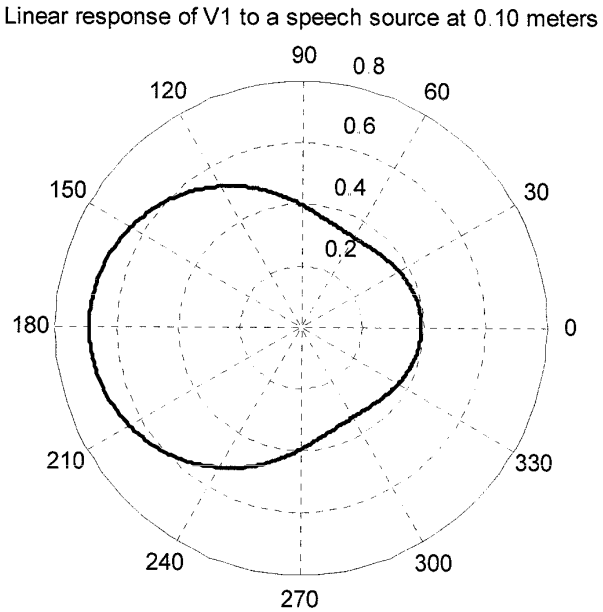

FIGURE 6

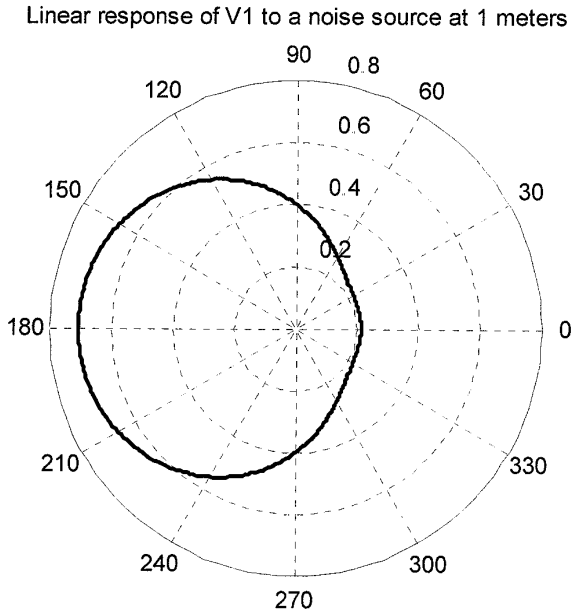

FIGURE 7

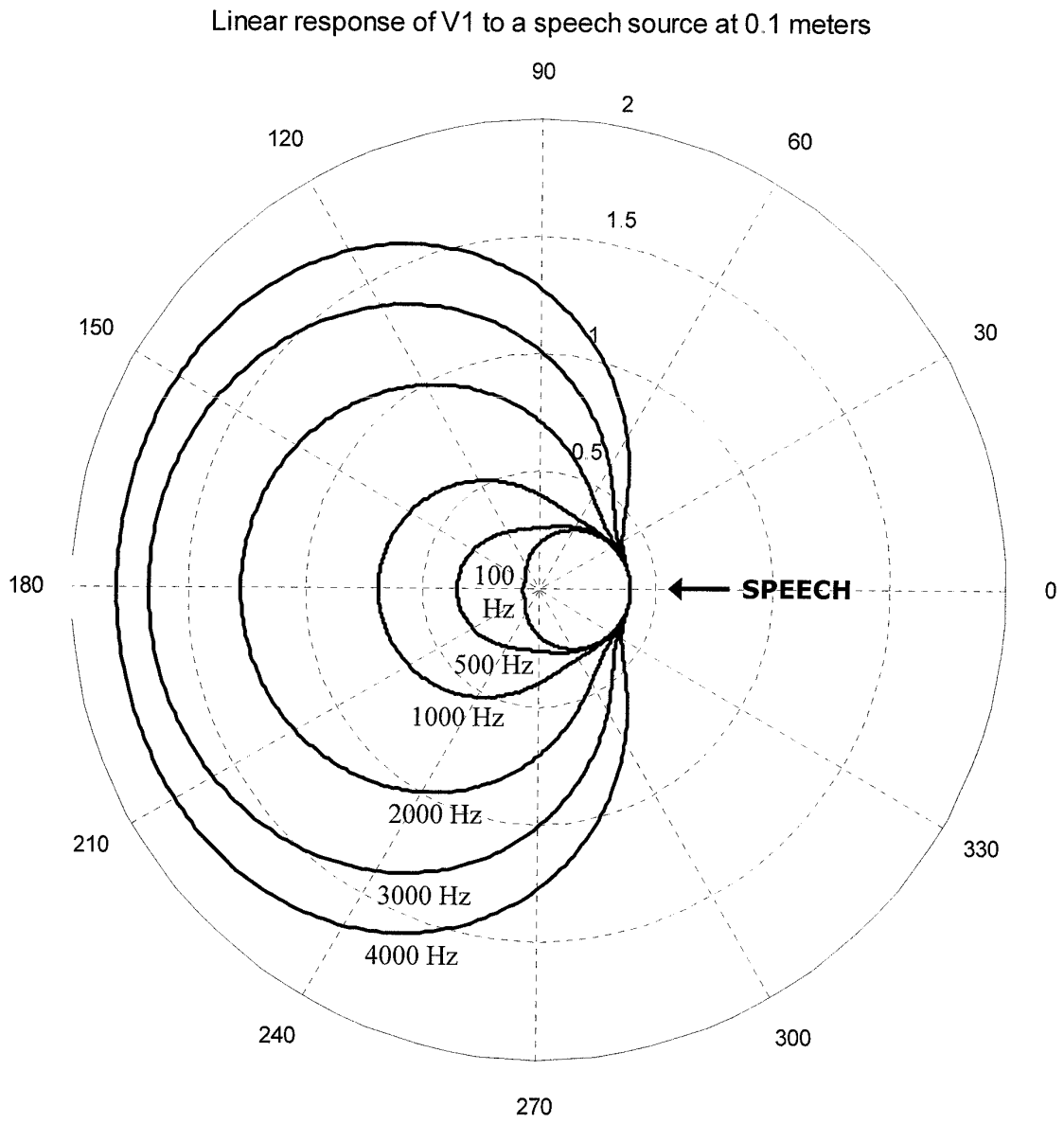
Linear response of V1 to a speech source at 0.1 meters
90
2
120
60
1.5
150
30
1
0.5
180
0
100 Hz
SPEECH
500 Hz
1000 Hz
2000 Hz
210
330
3000 Hz
4000 Hz
240
300
270

FIGURE 8

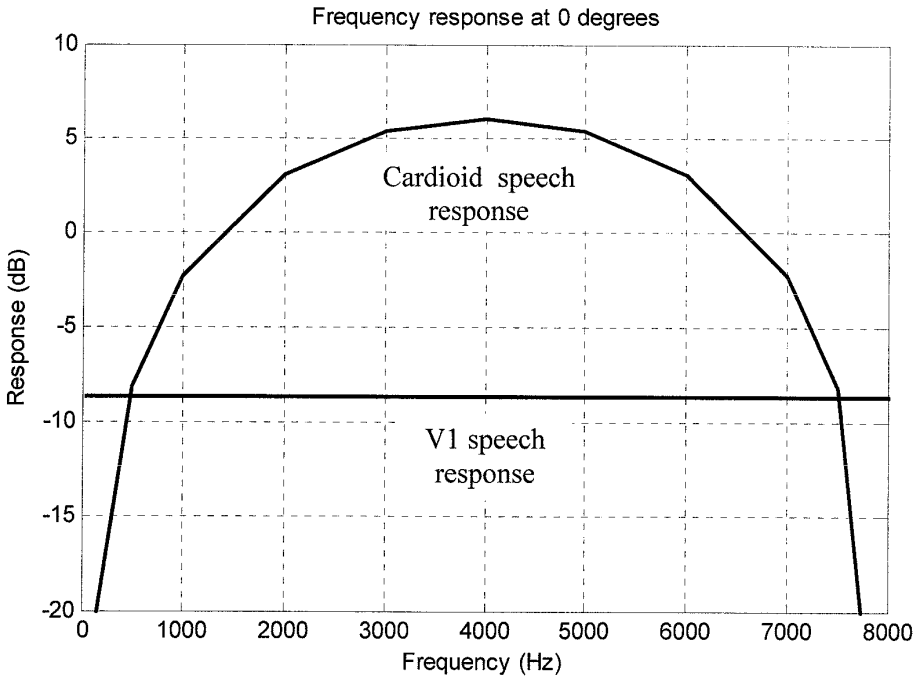
Frequency response at 0 degrees
Response (dB)
10
5
0
-5
-10
-15
-20
0
1000
2000
3000
4000
5000
6000
7000
8000
Frequency (Hz)
Cardioid speech response
V1 speech response

FIGURE 9

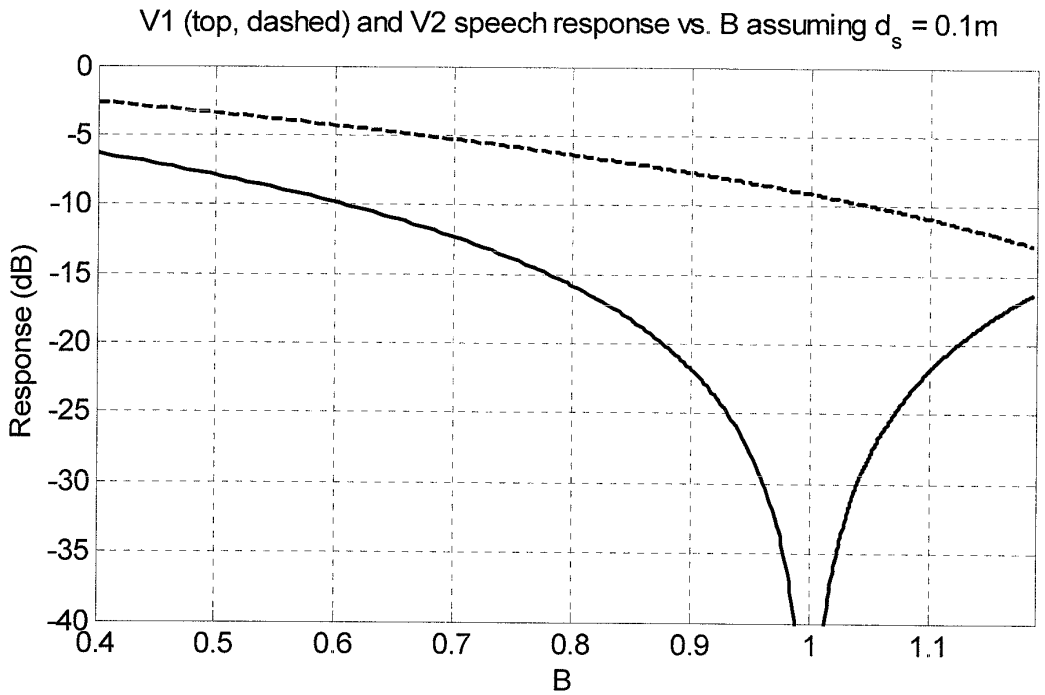

FIGURE 10

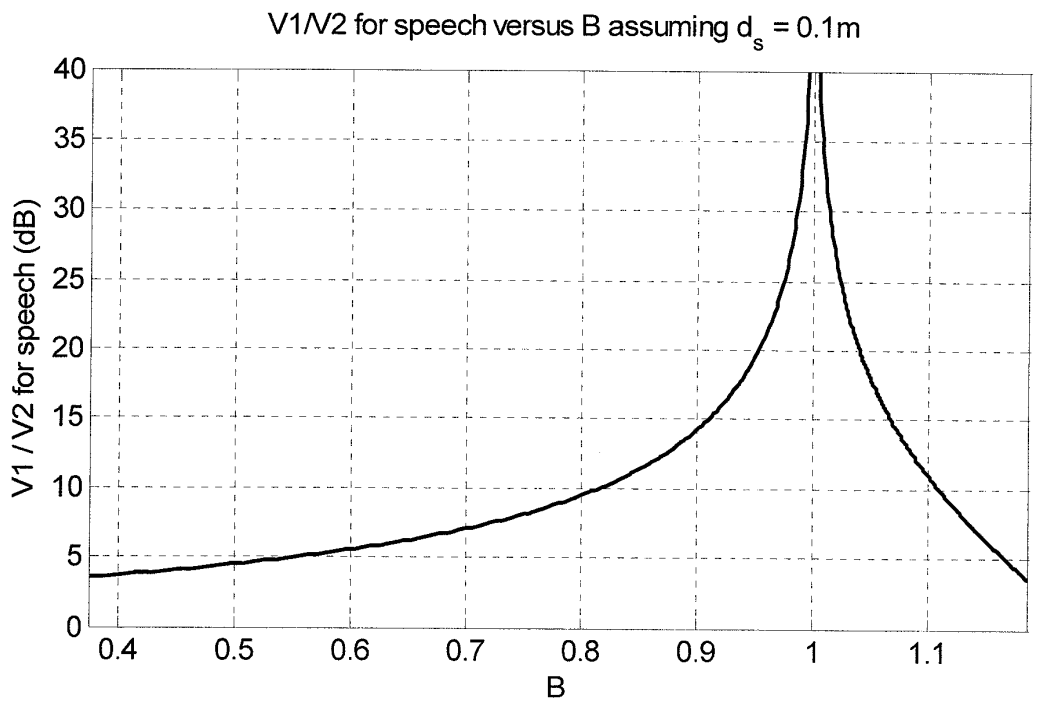

FIGURE 11

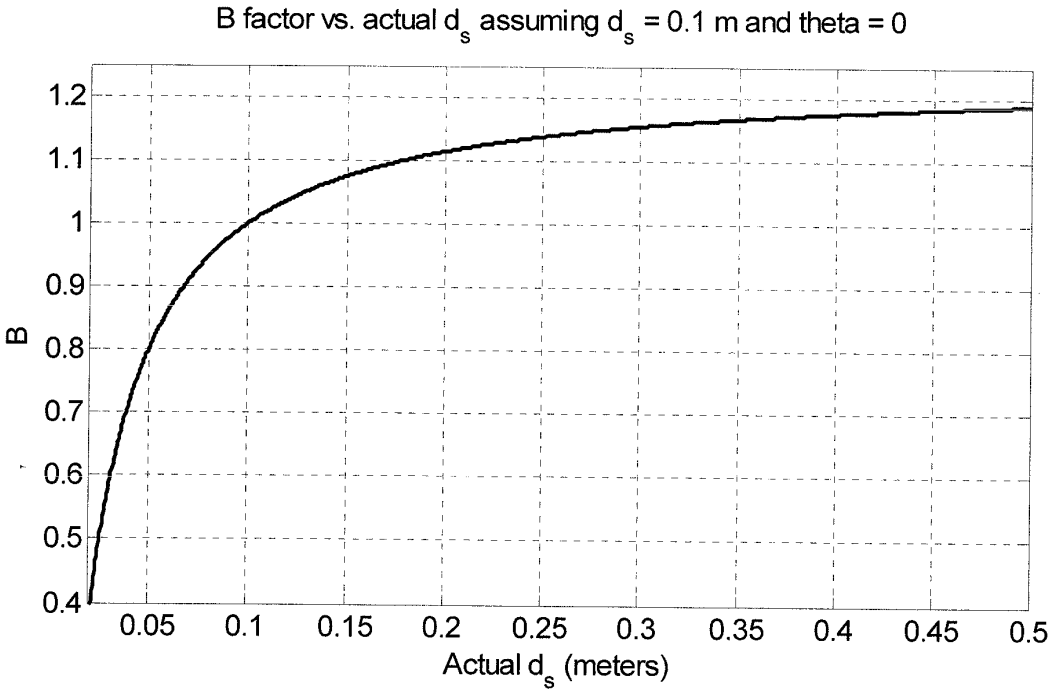

FIGURE 12

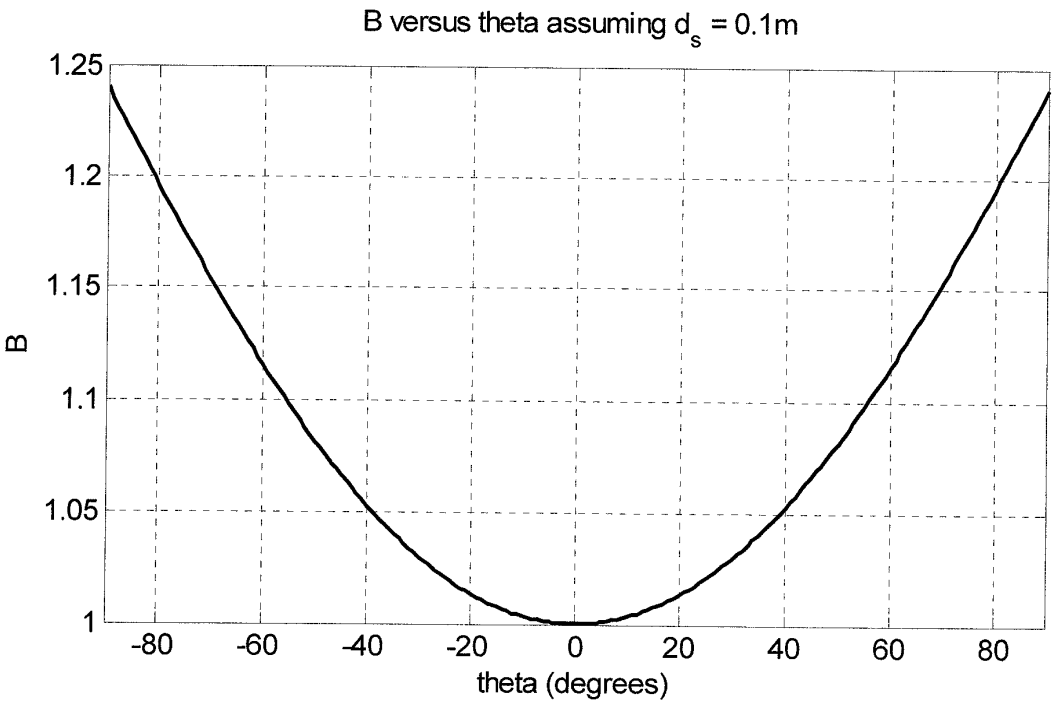

FIGURE 13

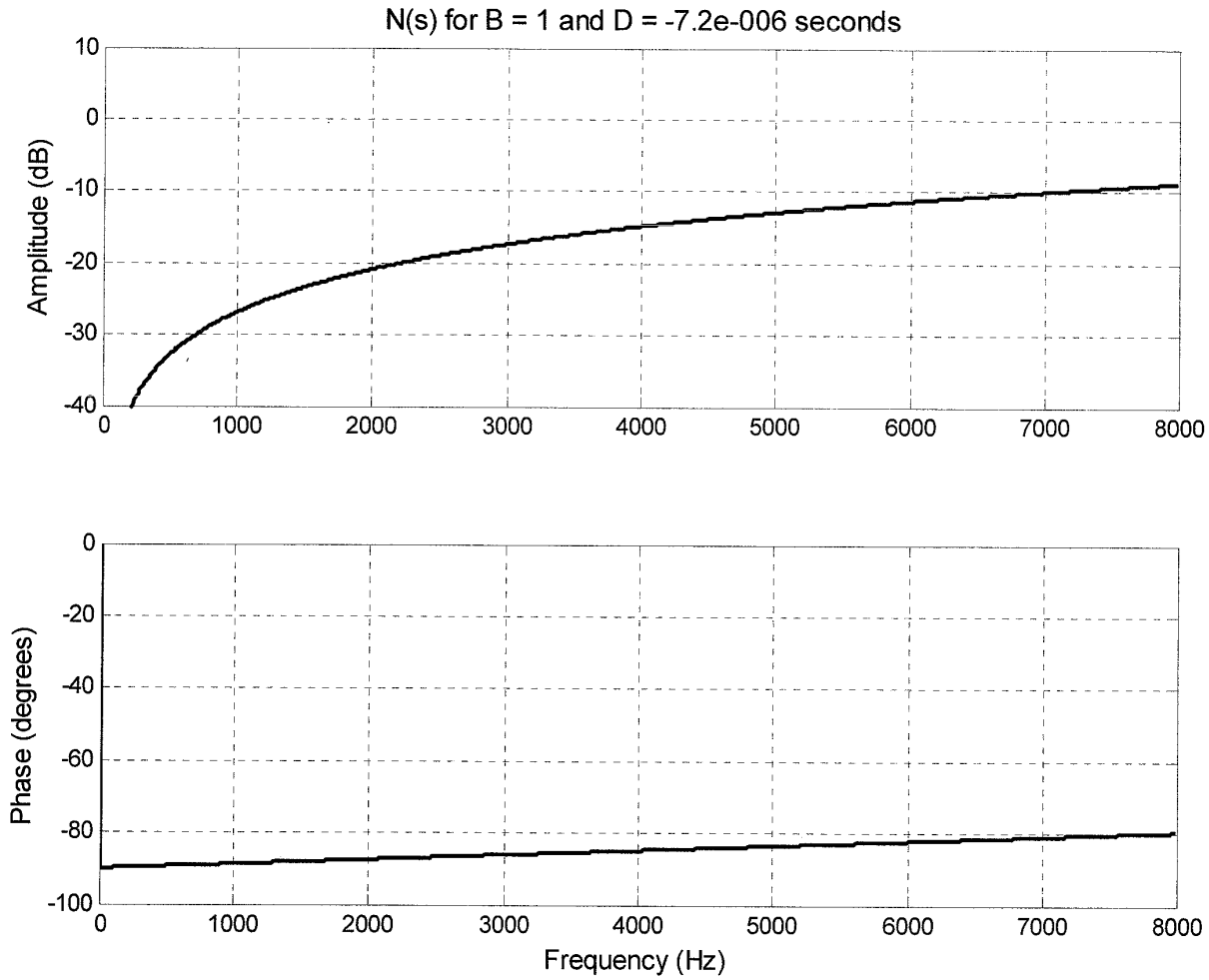
N(s) for B = 1 and D = -7.2e-006 seconds
Amplitude (dB)
10
0
-10
-20
-30
-40
0
1000
2000
3000
4000
5000
6000
7000
8000
Phase (degrees)
0
-20
-40
-60
-80
-100
Frequency (Hz)

FIGURE 14

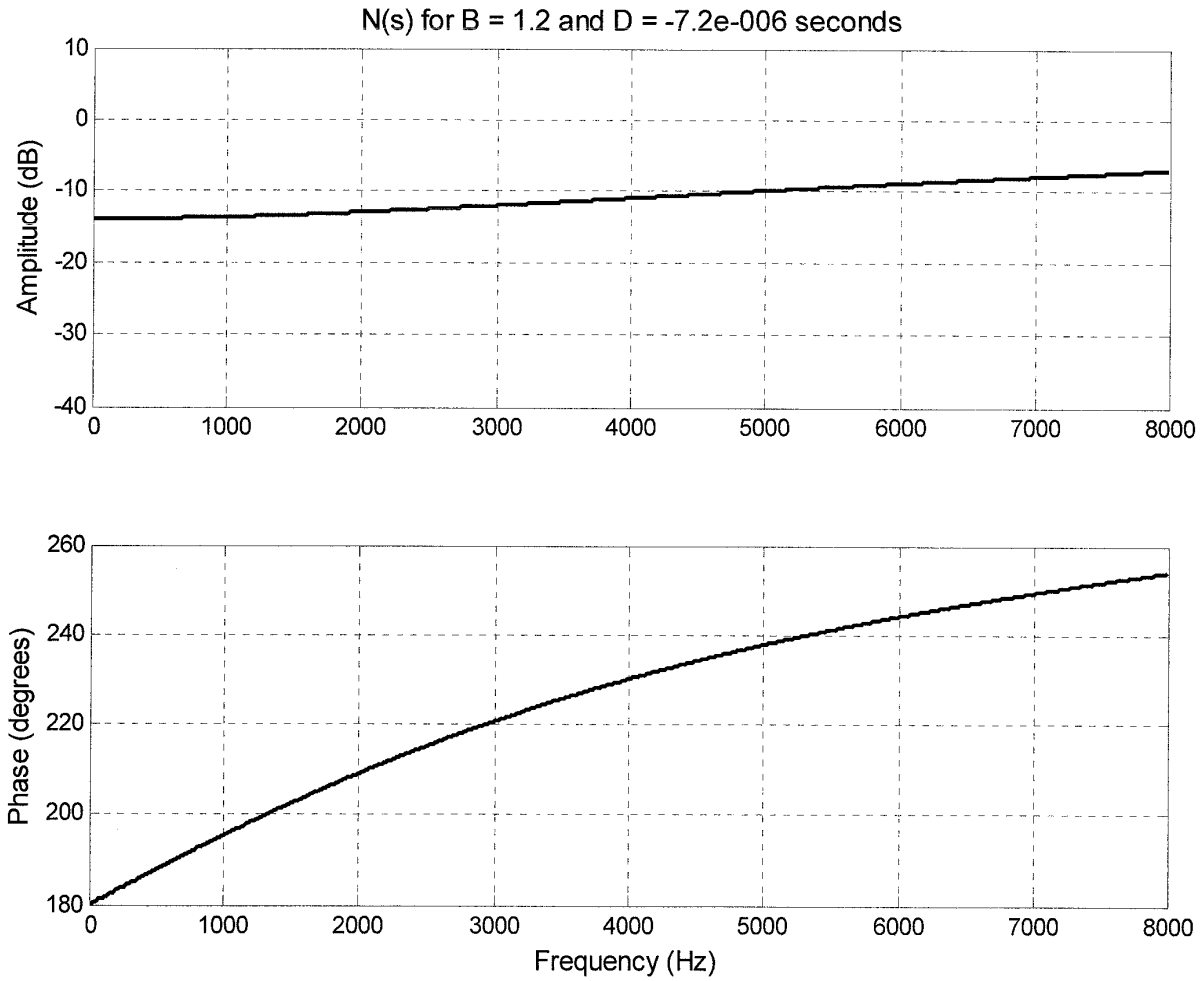
N(s) for B = 1.2 and D = -7.2e-006 seconds
Amplitude (dB)
10
0
-10
-20
-30
-40
0
1000
2000
3000
4000
5000
6000
7000
8000
Phase (degrees)
260
240
220
200
180
Frequency (Hz)

FIGURE 15

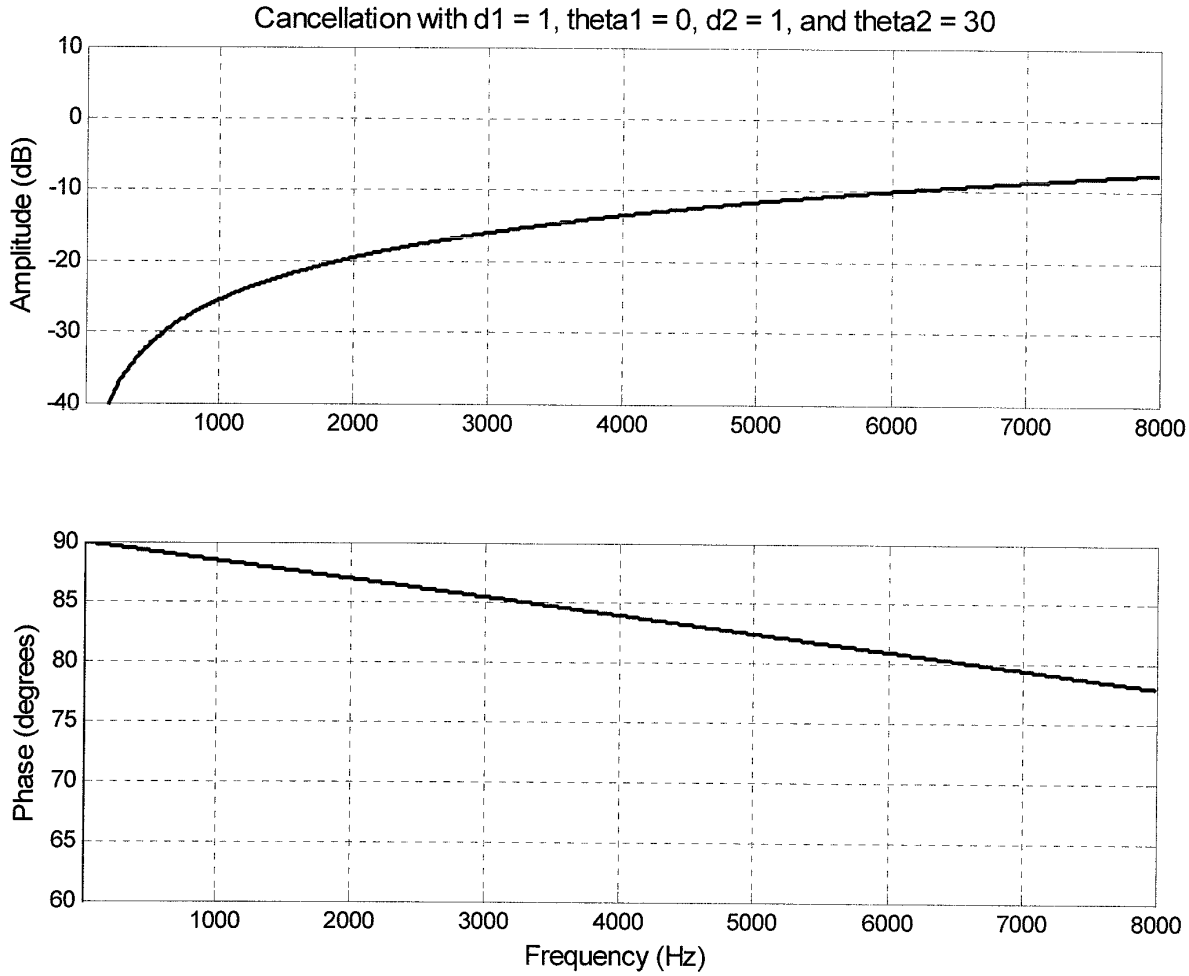
Cancellation with d1 = 1, theta1 = 0, d2 = 1, and theta2 = 30
Amplitude (dB)
10
0
-10
-20
-30
-40
1000
2000
3000
4000
5000
6000
7000
8000
Phase (degrees)
90
85
80
75
70
65
60
Frequency (Hz)

FIGURE 16

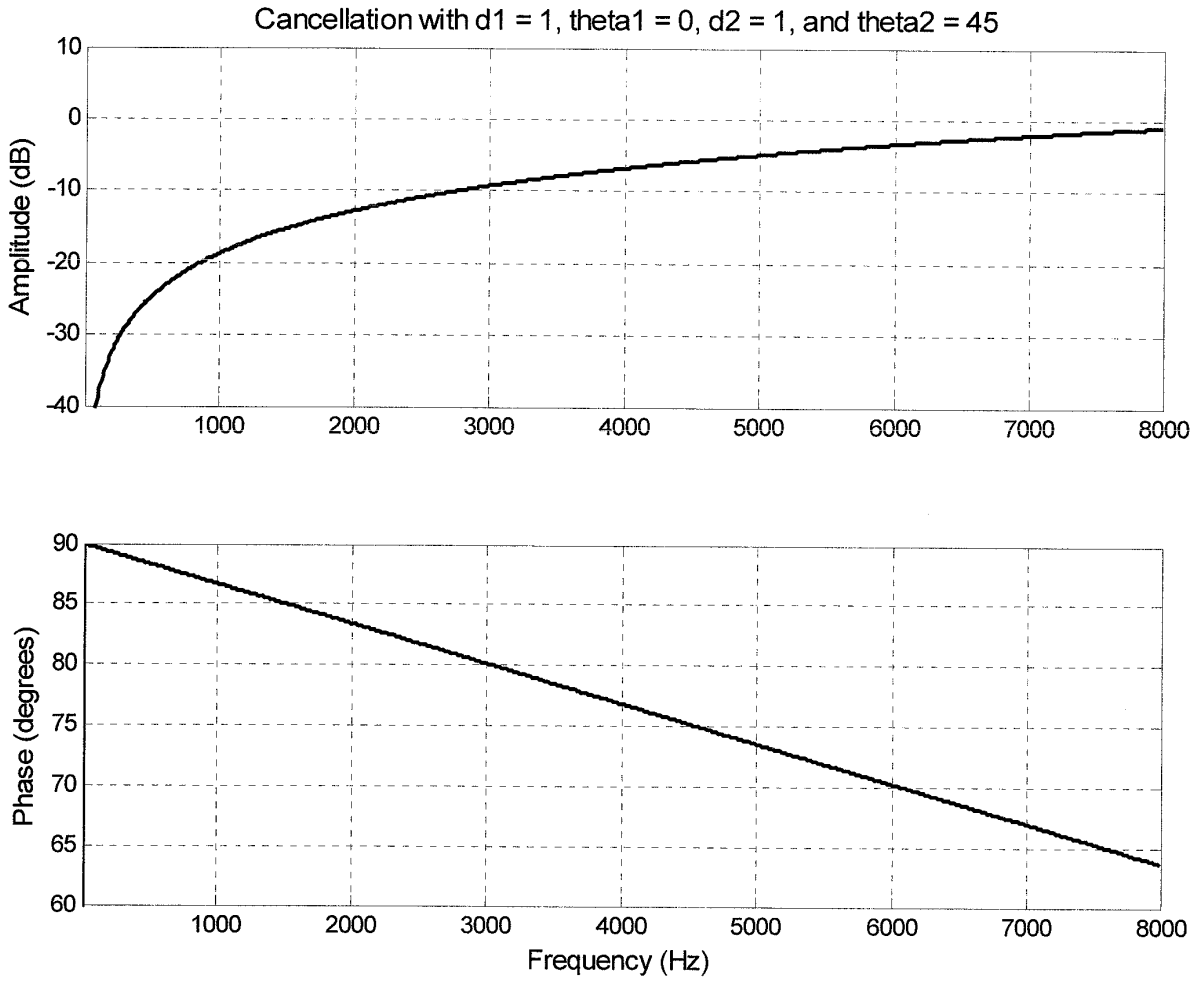
Cancellation with d1 = 1, theta1 = 0, d2 = 1, and theta2 = 45
Amplitude (dB)
10
0
-10
-20
-30
-40
1000
2000
3000
4000
5000
6000
7000
8000
Phase (degrees)
90
85
80
75
70
65
60
1000
2000
3000
4000
5000
6000
7000
8000
Frequency (Hz)

FIGURE 17

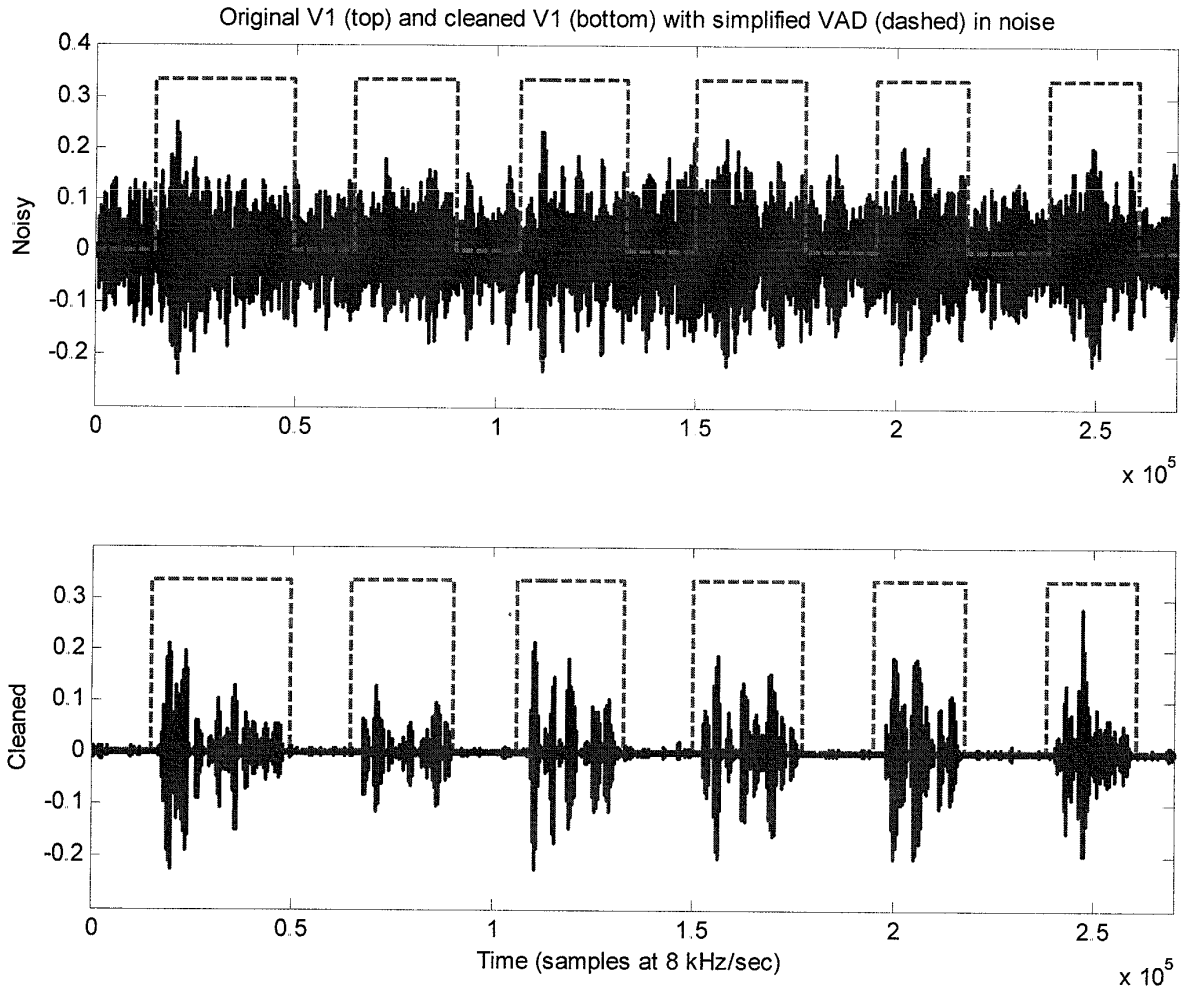
Original V1 (top) and cleaned V1 (bottom) with simplified VAD (dashed) in noise
Noisy
0.4
0.3
0.2
0.1
0
-0.1
-0.2
0
0.5
1
1.5
2
2.5
x 10^5
Cleaned
0.3
0.2
0.1
0
-0.1
-0.2
0
0.5
1
1.5
2
2.5
Time (samples at 8 kHz/sec)
x 10^5

FIGURE 18

Electronic Patent Application Fee Transmittal

Application Number:	
Filing Date:	
Title of Invention:	Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System
First Named Inventor/Applicant Name:	Greg Burnett
Filer:	Richard L. Gregory/Jerry Donnard
Attorney Docket Number:	ALPH.P035P2

Filed as Large Entity

Provisional Filing Fees

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
Provisional application filing	1005	1	210	210
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Miscellaneous:				
			Total in USD ($)	**210**

Electronic Acknowledgement Receipt

EFS ID:	3161083
Application Number:	61045377
International Application Number:	
Confirmation Number:	2839
Title of Invention:	Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System
First Named Inventor/Applicant Name:	Greg Burnett
Customer Number:	53186
Filer:	Richard L. Gregory/Jerry Donnard
Filer Authorized By:	Richard L. Gregory
Attorney Docket Number:	ALPH.P035P2
Receipt Date:	16-APR-2008
Filing Date:	
Time Stamp:	14:09:51
Application Type:	Provisional

Payment information:

Submitted with Payment	yes
Payment Type	Credit Card
Payment was successfully received in RAM	$210
RAM confirmation Number	7567
Deposit Account	
Authorized User	

File Listing:

Document Number	Document Description	File Name	File Size(Bytes) /Message Digest	Multi Part /.zip	Pages (if appl.)

1	Application Data Sheet	ADS_ALPHP032P2.pdf	1865018 75f68b7df9973129162411a96c8464414019adb4	no	4
Warnings:					
Information:					
2		Provisional_Application_ALPHP035P2.pdf	3142830 f710e6d39c370b98a74212fc51353204b08c9ef1	yes	43
	Multipart Description/PDF files in .zip description				
	Document Description		**Start**	**End**	
	Transmittal of New Application		1	1	
	Specification		2	30	
	Drawings-only black and white line drawings		31	43	
Warnings:					
Information:					
3	Fee Worksheet (PTO-06)	fee-info.pdf	8156 865c563f68f25d709b381bc307dd7b18c38db170	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	5016004		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

PTO/SB/14 (07-07)
Approved for use through 06/30/2010. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it contains a valid OMB control number.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	ALPH.P035P2
	Application Number	
Title of Invention	Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System	

The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76.
This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.

Secrecy Order 37 CFR 5.2

☐ Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)

Applicant Information:

Applicant 1 Remove

Applicant Authority ◉ Inventor ◯ Legal Representative under 35 U.S.C. 117 ◯ Party of Interest under 35 U.S.C. 118

Prefix	Given Name	Middle Name	Family Name	Suffix
	Greg		Burnett	

Residence Information (Select One) ◉ US Residency ◯ Non US Residency ◯ Active US Military Service

City	Dodge Center	State/Province	MN	Country of Residence i	US

Citizenship under 37 CFR 1.41(b) i: US

Mailing Address of Applicant:

Address 1	150 Executive Park Blvd., Suite 4550
Address 2	
City	San Francisco
State/Province	CA
Postal Code	94134
Country i	US

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. Add

Correspondence Information:

Enter either Customer Number or complete the Correspondence Information section below.
For further information see 37 CFR 1.33(a).

☐ **An Address is being provided for the correspondence Information of this application.**

Customer Number	53186	
Email Address	rgregory@csgip.com	Add Email Remove Email

Application Information:

Title of the Invention	Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System		
Attorney Docket Number	ALPH.P035P2	Small Entity Status Claimed	☐
Application Type	Provisional		
Subject Matter	Utility		
Suggested Class (if any)		Sub Class (if any)	
Suggested Technology Center (if any)			
Total Number of Drawing Sheets (if any)	13	Suggested Figure for Publication (if any)	

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	ALPH.P035P2
	Application Number	
Title of Invention	Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System	

Publication Information:

☐	Request Early Publication (Fee required at time of Request 37 CFR 1.219)
☐	**Request Not to Publish.** I hereby request that the attached application not be published under 35 U.S. C. 122(b) and certify that the invention disclosed in the attached application **has not and will not** be the subject of an application filed in another country, or under a multilateral international agreement, that requires publication at eighteen months after filing.

Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Enter either Customer Number or complete the Representative Name section below. If both sections are completed the Customer Number will be used for the Representative Information during processing.

Please Select One:	◉ Customer Number	◯ US Patent Practitioner	◯ Limited Recognition (37 CFR 11.9)
Customer Number	53186		

Domestic Benefit/National Stage Information:

This section allows for the applicant to either claim benefit under 35 U.S.C. 119(e), 120, 121, or 365(c) or indicate National Stage entry from a PCT application. Providing this information in the application data sheet constitutes the specific reference required by 35 U.S.C. 119(e) or 120, and 37 CFR 1.78(a)(2) or CFR 1.78(a)(4), and need not otherwise be made part of the specification.

Prior Application Status			Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)

Additional Domestic Benefit/National Stage Data may be generated within this form by selecting the **Add** button. Add

Foreign Priority Information:

This section allows for the applicant to claim benefit of foreign priority and to identify any prior foreign application for which priority is not claimed. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55(a).

Remove

Application Number	Country i	Parent Filing Date (YYYY-MM-DD)	Priority Claimed
			◉ Yes ◯ No

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button. Add

Assignee Information:

Providing this information in the application data sheet does not substitute for compliance with any requirement of part 3 of Title 37 of the CFR to have an assignment recorded in the Office.

Assignee 1 Remove

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	ALPH.P035P2
	Application Number	
Title of Invention	Generalized Omnidirectional Implementation of an Adaptive Noise Suppression System	

If the Assignee is an Organization check here. ☐				
Prefix	**Given Name**	Middle Name	**Family Name**	Suffix
Mailing Address Information:				
Address 1				
Address 2				
City		**State/Province**		
Country [i]		Postal Code		
Phone Number		Fax Number		
Email Address				
Additional Assignee Data may be generated within this form by selecting the **Add** button.				Add

Signature:

A signature of the applicant or representative is required in accordance with 37 CFR 1.33 and 10.18. Please see 37 CFR 1.4(d) for the form of the signature.					
Signature	/Richard L. Gregory, Jr./			Date (YYYY-MM-DD)	2008-04-16
First Name	Richard	Last Name	Gregory	Registration Number	42607

This collection of information is required by 37 CFR 1.76. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 23 minutes to complete, including gathering, preparing, and submitting the completed application data sheet form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

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5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

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8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.