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NOISE SUPPRESSION USING ACTIVE NOISE CONTROL AND MASKING

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Abstract

This paper attempts to make the public/workplace environment more pleasant by combining techniques of active noise control (ANC) and the limited ability of the human auditory system to register a sound in the vicinity of another sound of sufficiently higher intensity. The paper examines the limitations of ANC and masking techniques, and proposes a way to integrate these techniques. A process for the selection of a suitable masker and its subsequent integration with an ANC system will be introduced. A solution has been proposed to overcome the difficulties of achieving a reduction of the undesirable, as well as preserving/enhancing the desirable content in the audio domain. Several implementation issues will also be addressed in this paper towards lowering the annoyance level of engine noise.

1. INTRODUCTION

A common problem encountered in public or even private spaces is the incessant noise in one's surroundings. It is common to encounter construction equipment or engine noise nowadays in urban or even rural settings. The detrimental effects of persistent noise on human health [1] are well known and give us an impetus to find better ways for suppressing noise and making our surroundings pleasant. There is abundant literature available on the topic of active noise control (ANC) since the idea is not new and was patented by Leug in 1936[2]. A basic block diagram of the system [3] using an adaptive filter to control the secondary source output is illustrated in Figure 1(a). A basic narrowband feed forward ANC makes use of the fact that many noises, such as those generated by engines, compressors, motors, fans and propellers are periodic in nature and can use a sensor other than a microphone for providing the reference input.

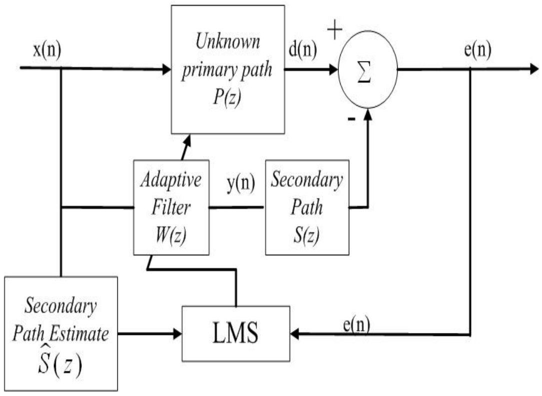

Figure 1(a) : ANC using an adaptive filter

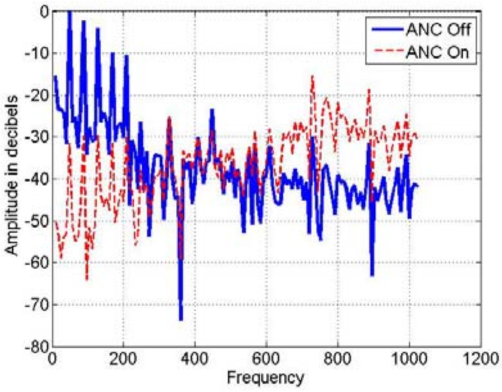

Figure 1(b):Engine noise ANC simulation

In practical applications, periodic noise contains harmonics in addition to its fundamental. Further the effectiveness of ANC systems above 500 Hz falls off significantly [4].Figure 1(b) illustrates the effect of an ANC simulation on a noise source. It is clearly evident that even in a controlled simulation ANC is not fully effective. There is a marked deterioration in the performance beyond 500 Hz. This limitation provides the motivation for finding an alternative solution. As the current ANC systems are not able to completely decimate the unwanted noise, the residual noise is sought to be masked by a pleasant sound in the least intrusive manner possible.

This paper is organized as follows. In the next section, the masking effect and its implications in everyday life are explained. This includes a rudimentary explanation of psychoacoustic model–I (used in MPEG audio coding [5]) employed for applications which do not have a rigorous requirement of accuracy and prefer computational simplicity. Section 3 describes the process used for determining the suitable maskers from an available pool and their relative masking score. Section 4 introduces the integrated ANC and masking system proposed in this paper. Section 5 concludes this paper and suggests a few possible applications of the proposed system.

2. THE PSYCHOACOUSTICAL MODEL AND ITS APPLICATIONS

The psychoacoustic model and its exploitation for the present requirement are briefly discussed in this section. The psychoacoustic model [5]-[10] seeks to imitate the human auditory mechanism to a reasonable extent in order to discard the redundant information or exploit this information in any other useful manner. The primary objective in this paper is not to discard the redundant information but to find out the amplitude levels below which, another signal even if present, is imperceptible. The auditory model processes information to find out the final masking threshold. The first step is to translate the actual audio frame signal into the frequency domain using the Fast Fourier Transform. In the frequency domain, the power spectrum energy per critical band and the spread energy per critical band are calculated to estimate the masking threshold. Finally, frequency domain output is translated back into its time-domain representation and the next frame is processed. The human auditory system is not adequately represented by the linear frequency scale. A more realistic scale is the Bark scale [11].

The brain processes information in spatial-frequency segments or regions known as critical bands. By definition, a Bark unit has the width of one critical band. The first step in the frequency domain is to calculate the power spectrum of the incoming signal by summing the squares of the real and imaginary components.

Subsequently, the energy of each critical band or Bark is calculated by summing up all the power spectral components in each band. These bands' energy is used to compute the spread masking threshold. By convoluting the basilar membrane spreading function and the energy per critical band, the spread masking across critical bands is calculated. This is used to calculate the masking threshold in each critical band [5]. However, the calculation of masking index based upon the sound being noise-like or tone-like is also essential. This is because noise has a greater masking ability compared to a tone. In most audio applications, the signal is further processed to discard the imperceptible contents, adding of digital fingerprints, embed additional information or encrypted and then reconverted to the time domain. However, the same is not required here. The objective in this work is just to estimate masking thresholds of relevant critical bands.

3. SELECTION OF THE BEST MASKER FROM A COLLECTION OF POSSIBLE MASKERS

As an estimate of the noise of any ANC system can be available, a process for selection of the best possible masker is outlined below. The three basic conditions for an effective masker are:

(I) The underlying noise should be transparent to an average listener.

(II) The masker should be played at the lowest volume possible.

(III) The masker should sound pleasant to the listener.

A psychoacoustic processor is used to objectively rank a collection of potential maskers. The process of ranking the maskers is explained in the next paragraph. Although this objective measurement may fulfil the first two conditions, the third condition is based upon subjective assessment and may differ from person to person.

The psychoacoustic processor first takes in the estimated noise to be masked, calculates its sound pressure level (SPL) and stores its values (Figure 2a). Next, each potential masker is compared with the estimated noise signal. It calculates the masking threshold for all relevant bands. Figure 2(b) [7] illustrates the global masking threshold (red dashed line) of the masker SPL(blue solid line). The brown solid line displays the absolute hearing threshold. The global masking threshold of the masker is compared with the noise SPL.

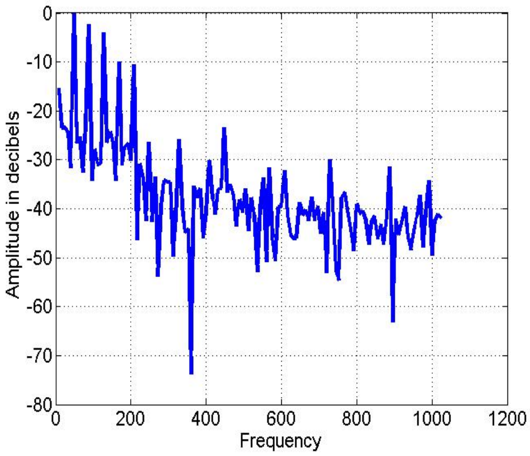

Figure 2(a) : Maskee SPL values

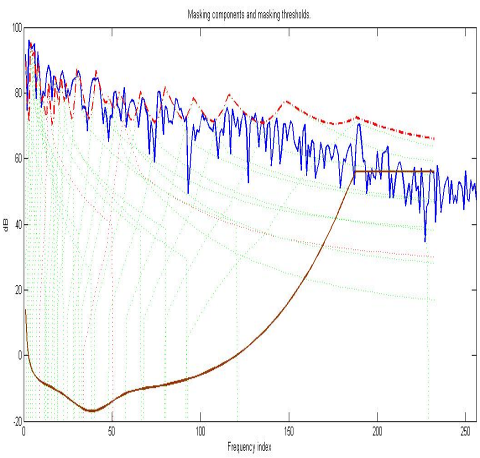

Figure 2(b): Masker threshold values

For example, let us consider *n* frequency points of interest and their amplitude levels for the masker and the maskee. For, these n points of both the signals, we can calculate the sum of differences in amplitude when at any one point, the maskee SPL and the masker threshold are exactly coinciding and rest of the threshold points are above their corresponding maskee SPL points. The energy for each masker is then calculated, keeping the maskee amplitude constant. The masker with the least energy is ranked first, the second lowest energy ranked next, and so on for all available maskers.

Let $X = (x_1, \ldots, x_n)$ be the maskee with its *n* frequency amplitudes.

and $Y_j = (y_1, \ldots, y_n)$ be a potential masker with corresponding *n* frequency amplitudes..

With the set of all potential maskers being $[Y_1, \ldots, Y_m]$

The matching process for Y_j and X is as follows:

Step 1: Scale down the amplitude of Yj till Max $(Y_j) < Min(X)$

Step 2: $D_j = Y_j - X$

Step 3: Increase gain of Y_j *by Max* $(|D_j|)$

Step 4: Again calculate $D_j = Y_j - X$

Masking Score $S_j = \sum_{i=1}^{n} d_i$ *where* d_i *is the* i^{th} *component of* D_j

This value is the masking score of the masker Y_j. Sorting the values of the masking scores in ascending order provides us with a ranking of all available maskers. Only the selected masker signal and the amplification required are passed on to the system for real-time processing. A small amplification of 1 or 2 dB can be optionally added to the selected masker to produce a better result.

4. INTEGRATED ANC-MASKING SYSTEM

Although systems integrating ANC and psychoacoustic processing have been implemented before, they have been mostly used in systems where the noise and audio are processed electrically [12], [13]. Once the masker audio (desirable sound, such as soothing music or sounds from nature) and the noise (undesirable, such as engine noise) are out in the acoustic domain, the complexity of combining ANC and masking is raised to a higher level. An application of this method has been used in [14] to cancel out snore noise and mask the residual with a pleasant noise to minimize disturbance to the snorer's partner. This paper extends the work in [14] using ANC and masking techniques for handling narrowband engine noise, with non-stationary characteristics. Narrowband ANC has been implemented in many systems [12], [13]; however it has certain limitations such as a perceivable residual noise. Further the control is not effective for higher frequencies [4] above 500 Hz.

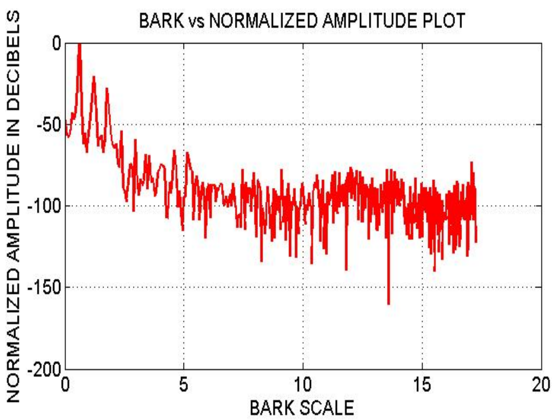

Figure 3(a): Engine Noise in Bark scale

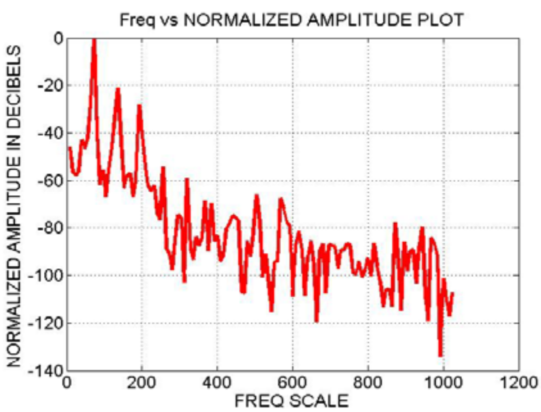

Figure 3(b): Engine Noise in Hz

To keep the implementation simple, we consider an engine noise which exhibits narrowband harmonics. Figure 3(a) illustrates the spectrum of the recorded engine noise in a Bark scale. The narrowband nature of the signal is further highlighted in Figure 3(b) which covers a limited frequency range. It can be observed from these figures that considerable attenuation can be achieved in the sound pressure levels just by the reduction of the fundamental noise and its harmonics.

4.1 Secondary Path Modelling

Most available control algorithms for ANC require identification of the secondary path. The summing junction in Figure 1(a) represents acoustic superposition in the space from the canceling loudspeaker to the error microphone, where the primary noise is combined with the output of the adaptive filter. Therefore, to get a stable adaptive filter, it is necessary to compensate for the secondary-path transfer function which includes the digital-to-analog (DAC), reconstruction filter, power amplifier, loudspeaker, acoustic path from loudspeaker to error microphone, error microphone, preamplifier, anti-aliasing filter, and analog-to-digital converter (ADC). Hence, this is a vital component of any ANC system. Figure 4 illustrates the technique for modelling the secondary path. A known signal u(n) is passed through the unknown secondary path and the adaptive filter is updated iteratively so that it resembles the actual secondary path.

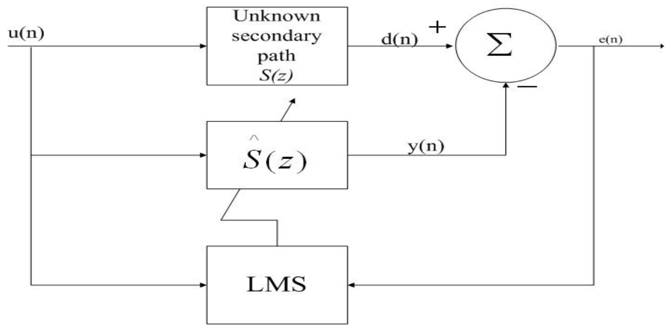

Figure 4: Secondary Path Modelling

The magnitude and phase responses of the actual and modelled secondary path shown in Figures 5(a) and 5(b) illustrate that a reasonably good estimate can be obtained using the LMS algorithm. Usually white noise is used as the training signal as it covers a wide spectrum and the actual noise sought to be cancelled may not be available. As white noise sounds like a hiss and is unpleasant to the ear, the selected masker is used for the training. While any adaptive FIR filtering algorithm could be used for designing the secondary propagation path estimate, the normalized LMS algorithm is often used due to its simplicity and robustness [4]. We can observe from Figures 6(a) and 6(b) that even though the error is initially larger for the system using pleasant sound instead of white noise, after about 0.08 seconds the error is comparable. Furthermore, the error seems to be converging quickly. Thus, a pleasant masker can be used without any significant reduction in the system performance.

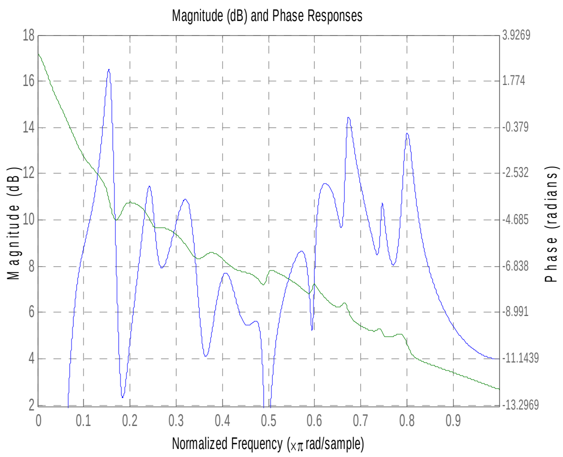

Figure 5(a): Actual Secondary Path

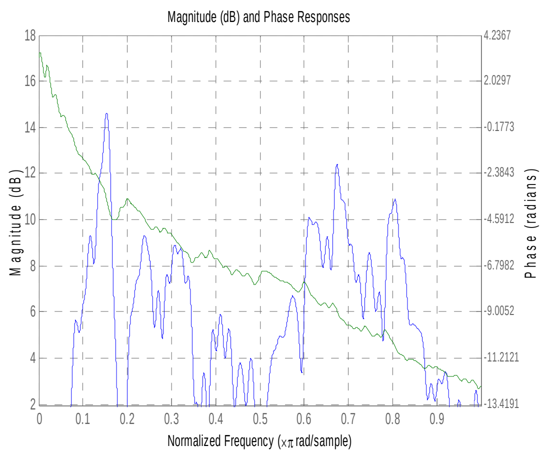

Figure 5(b): Modelled Secondary Path

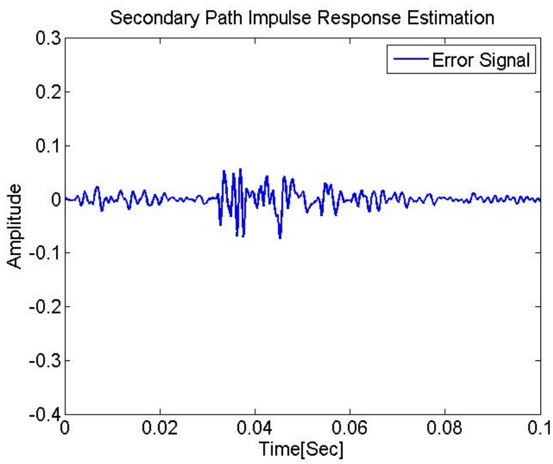

Figure 6(a): Modelling error using white noise

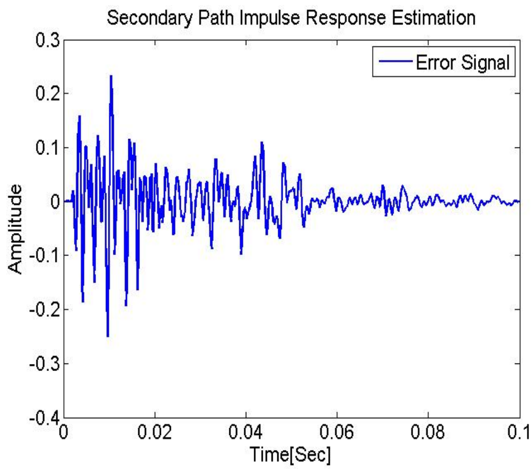

Figure 6(b): Modelling error using masker

4.2 Integrated ANC and Masking:

It can be seen from Figure 7 that the integrated ANC and masking system is different from the conventional ANC systems. The dashed box represents the flow of the signals in the acoustic domain (open environment). The objective here is to ensure that the sound obtained after the superposition, in addition to being minimized resembles audio more than the noise. Equation 1 illustrates the superposition effect at the error microphone.

$$e(n) = y'(n) + d(n) \qquad (1)$$

The outcome of the superposition of the control signal after the secondary path and the primary disturbance should sound like music to the human ear. To achieve this objective we compare it with the desired signal.

$$e'(n) = e(n) * p_2(n) - g.\ r(n) * \hat{p}_2(n) \qquad (2)$$

Here $r(n)$ is the masker, $P_2(z)$ denotes the effect of the electrical path from the error microphone to the comparison circuit and $\hat{P}_2(z)$ is a filter that compensates for this effect. The gain constant g for the masker is explained in section 4.3.The symbol '*' denotes convolution and '.'denotes multiplication. The residual noise *e'* (*n*) is further processed in a manner identical to a conventional ANC system [4]. Similarly, the combined audio and noise reference input is compared to the recorded audio and the difference should be the noise. This noise is filtered through an estimation of the secondary path and is used to update the adaptive filter *W* (*z*) by the least mean square (LMS) algorithm.

$$w(n+1) = w(n) + \mu . e'(n). \{x(n)-g.r(n)\} * \hat{s}(n) \qquad (3)$$

One major problem faced here is that the sampling frequency used in a narrowband ANC, is quite low (a few kHz). However, any pleasant noise covers a gamut of frequencies and ceases to be pleasant the moment it is restricted to a very narrow band. Thus, there is a mismatch between the sampling frequency requirements of the noise and the masker. A solution suggested in [12] is to equalize the sampling rate by up sampling ANC part or down sampling the masking portion. However, the subjective effects of this course of action and its effect on the computational speed remain to be assessed.

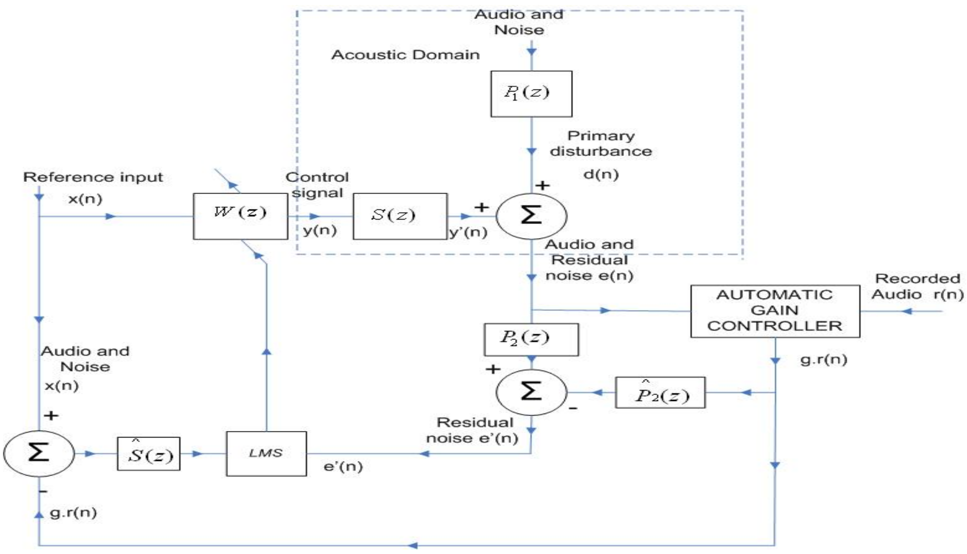

Figure 7: Integrated ANC-Masking System

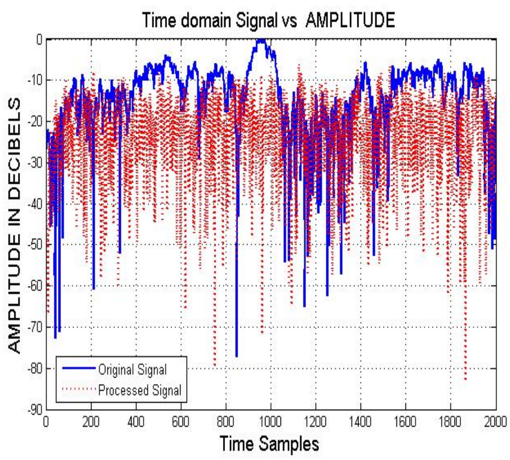

Figure 8(a): Amplitude reduction

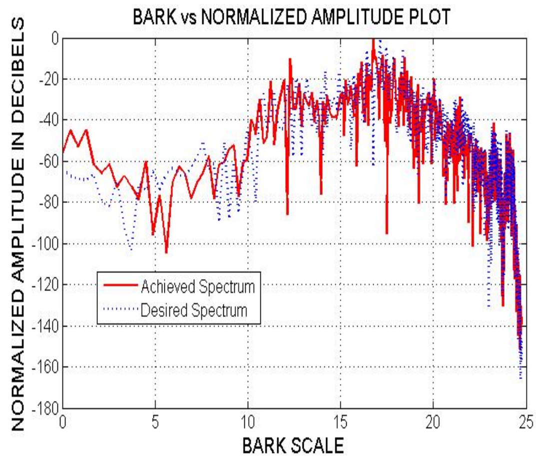

Figure 8(b): Spectrum similarity

As can be seen from Figure 8(a) the overall envelope of the undesirable noise has reduced. It can be observed that up to a maximum of 15 dB reduction has been achieved, and on average, there is nearly 10 dB reduction. One interesting characteristic is that the reduction is more at the peaks of the original signal and less when the original amplitude is low. Figure 8(b) illustrates the spectral similarity between the desired output and the processed output. We can see that the similarity extends throughout the entire human auditory range. This is mandatory if we are to achieve a pleasant auditory experience. Furthermore, we can see that the similarity is less in the lower bands and more towards higher frequencies. This property is beneficial as we have seen earlier that ANC is more effective at lower frequencies. In contrast, with this technique we see that the performance gets better at higher frequencies. Hence we have achieved the twin objectives of reducing the undesired noise, as well as managing the residuals so that they sound pleasant.

4.3 Time-varying Maskee (Noise Source):

The solution for a narrowband, near stationary engine noise has been discussed so far. However, in a practical scenario, the engine revolutions per minute (rpm) is likely to vary, causing a change in the characteristic of the maskee (or noise) [15]. Hence, there is a need for the system to adapt to any such variation. The spectrum plots of the same engine noise at two rpms are shown in Figures 9(a) and 9(b). We can infer that the difference in the maskees may result in different maskers being found suitable.

To reduce the complexity of changing masker when the noise source is changing, we propose to perform automatic gain control of the selected masker levels so that condition I stated in Section 3 remains satisfied. We also assume that general trends/characteristics of noise source are not changing too drastically. After the ANC has converged to the new rpm residual noise, the spectrum of the last k samples of the error signal $e(n), e(n-1) \ldots\ldots e(n-k)$ is taken. Let the spectrum of the error signal and recorded audio be represented by $E(\omega)$ and $A(\omega)$, respectively. $E(\omega)$ contains both the recorded audio and residual noise components. So, the difference of $E(\omega)$ and $A(\omega)$ gives us the residual noise spectrum $M(\omega)$.

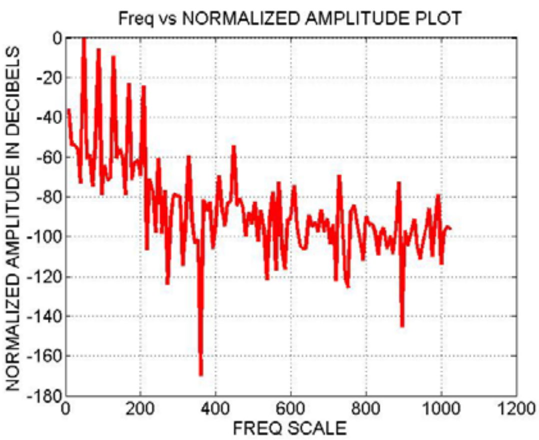

Figure 9(a) Spectrum at initial rpm

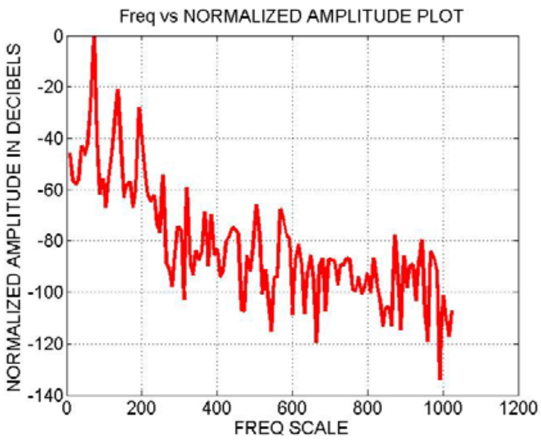

Figure 9(b) Spectrum at changed rpm

We test the above proposal for the two different maskees of Figure 9(a) and 9(b).The value of constant g is calculated using the algorithm outlined in Section 3 using $A(\omega)$ as the masker and $M(\omega)$ as the maskee. The signal flow is illustrated in Figure 10. It is pertinent to mention that this is not a continuous process and the change in gain takes place only on receiving an external trigger such as an electronic signal indicating a different rpm of the engine.

The effect of the amplification/attenuation on ANC and output spectrum quality is observed. On comparing Figures 11(a) and 11(b), it is apparent that the amplitude reduction due to ANC is nearly the same in both cases. Comparison of Figures 12(a) and 12(b) reveals that the lower frequencies are deviating slightly more from the desired spectrum in the case of changed maskee. However, the higher frequencies have almost similar performance in both cases. The similarity can be further evaluated using subjective tests. Hence, a reasonable solution for adapting to a change in the maskee has been obtained.

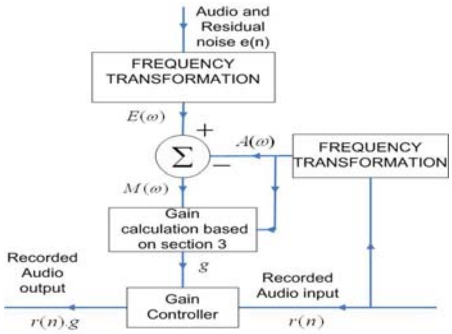

Figure 10: Signal flow for Automatic Gain Control of Masker

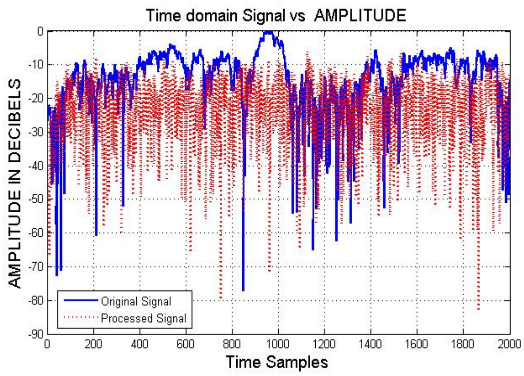

Figure 11(a): Amplitude reduction with original maskee

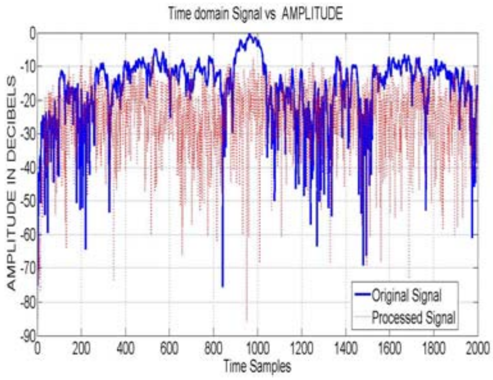

Figure 11(b): Amplitude reduction with changed maskee

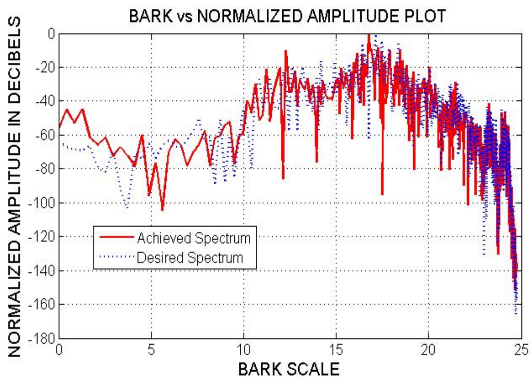

Figure 12(a): Spectrum comparison with original maskee

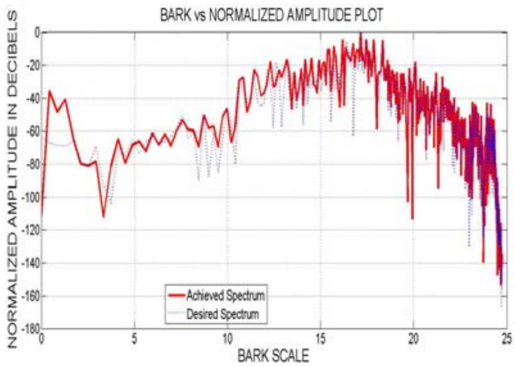

Figure 12(b): Spectrum comparison with changed maskee

5. CONCLUSION

We have presented an innovative implementation, combining the well researched techniques of ANC and psychoacoustic processing, in this paper. In addition to attenuating an unpleasant noise, the residual was managed in such a manner as to render it pleasant to the human ear using the masking effect. AGC was also exploited for taking care of any changes in the engine noise (unpleasant noise) by raising/attenuating the recorded audio (pleasant). The implementation can be used for myriad applications in the noisy environments encountered commonly these days. Particular applications such as in aircrafts, public transport, workplaces, restaurants and shopping malls can find ready use of the solution presented in this paper.

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