A Novel Approach to Bass Enhancement in Automobile Cabin

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ABSTRACT

An audio enhancement system is presented to improve the sound quality, especially the bass reproduction in the automobile cabin. High quality audio reproduction in cabin can be difficult due to a number of factors, including the noise in the cabin, performance, size of the loudspeakers in the car. The proposed system uses digital signal processing techniques with the built-in car audio system to tune and make use of the engine noise. The main problems are the tracking of the engine noise and tune the noise to match the audio signal. In this paper, we propose a solution based on frequency sampling filters to track and extract the audio signal, and a multi-frequency active noise equalizer to tune the engine noise. The results showed that the proposed system could be a promising approach for solving the bass audio reproduction, together with the noise control problems in automobile.

1. INTRODUCTION

Noise control and the high quality bass reproduction in automobile cabin are two interrelated problems. The later one can be difficult due to the high noise level present inside the car and quality of the loudspeakers. Traditional passive noise control techniques are only efficient at high frequencies. For the annoying low frequencies engine noises, passive techniques become costly and large sized, which are not suitable for use in automobile. Due to the efficiency of reducing low-frequency noise, the active noise control (ANC) [1] technique has received much attention since 1980s. The use of active noise control (ANC) in the automobile has been commonly reported, however few real applications were carried out on the mass production line [2].

On the other hand, with the development of multimedia digital signal processing (DSP) technologies, high-quality audio reproduction is becoming possible in the automobiles. However, there are existing challenges in reproducing high-quality bass in the car, due to the limited space and acoustic properties of most automobiles, and the low-frequency noise present in the cabin.

The ANC techniques generally produce good performance in cancelling the in-cabin narrowband engine noise. However it does not offer complete control over the engine noise. As in some practical applications, it is desired not only to retain some amount of noise, but also enhance it to extract important sound information of the system. For example, the driver may want to know how the engine is working when driving. Due to its flexibility of amplifying or attenuating noises at certain frequency components, active noise equalizer (ANE) [3] systems have potential applications in such cases.

In order to perform noise control for bass enhancement, we proposed a novel method that uses frequency sampling and multi-frequency ANE [3] to tune the engine noise, and convert the annoying low-frequency noise into useful audio components.

The remainder of this paper is structured as follows. Section 2 briefly presents the narrowband ANE system, followed by a description of the proposed system in Section 3. Results are given in Section 4 with conclusions in Section 5.

2. NARROW BAND ACTIVE NOISE EQUALIZER

The single-frequency narrowband ANE [3] system is based on an adaptive notch filter using FXLMS [1] algorithm. As shown in Figure 1, the secondary output is split into two branches: the cancelling branch and the balancing branch. A pseudo-error $e'(n)$ is used to trick the adaptive filter to converge to a desirable state, where the pseudo-error can be expressed as,

$$e'(n) = d(n) - y(n) * s(n) . \tag{1}$$

After convergence, the pseudo-error converges to zero. However, the actual residual noise $e(n)$ converges to,

$$\begin{aligned} e(n) &= d(n) - (1-\beta) y(n) * s(n) \\ &\approx \beta d(n), \end{aligned} \tag{2}$$

where β is known as the gain factor.

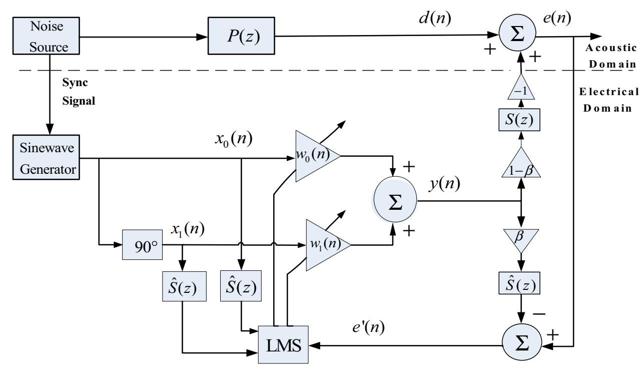


Fig. 1. Block diagram of single-frequency ANE system

Depending on the gain factor β, ANE can be classified into four working modes [3]:

(i) Cancellation mode ($\beta = 0$): In this mode, ANE functions as the conventional narrowband ANC.

(ii) Attenuation mode ($0 < \beta < 1$): The amount of attenuation is determined by factor β. So it is possible to retain some portion of the noise signal at the frequency of the reference signal.

(iii) Neutral mode ($\beta = 1$): The ANE system has no effect as there is no attenuation.

(iv) Enhancement mode ($\beta > 1$): The ANE functions as an amplifier that intends to enhance the noise at the frequency of the reference signal.

3. PROPOSED SYSTEM

The proposed system can be divided into three sub-systems: First, extracting the bass component from car audio system; second, post processing of the bass component audio signal; and third, devising a multi-frequency ANE system. A basic system diagram is shown in Figure 2.

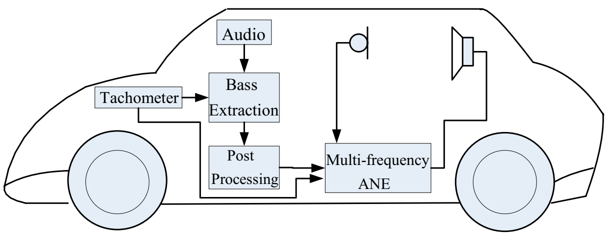


Fig. 2. System diagram

3.1. Audio bass component extraction

The components in the audio signal that we are interested to enhance are those close to the engine noise components. They are related to the engine revolutions per minute (RPM). As the engine RPM is not a constant value, the engine noise components change accordingly. So the filters must be able to self-reconstruct according to the engine RPM. to extract the desired audio signal. In other words, the filter's center frequency should be tuned by the engine RPM and on the predominant engine noise frequencies.

As shown in Figure 3, the audio signal is passed through a low pass filter with a cut off frequency at 700Hz, and the audio signal is decimated at a lower sampling frequency to 1.5 KHz. Therefore, lower computational load for processing bass information of the audio signal is used. After the decimation, we perform frequency sampling filtering (FSF)[4] to extract the desired frequency components. Depending on the number of predominant engine noise orders of the target engine, we use same the same number of FSF channels, each FSF channel corresponds to one engine noise order. A unique attraction of the FSF structure is that it allows recursive implementation of FIR filters, leading to both computational efficiency and fast on-line reconstruction.

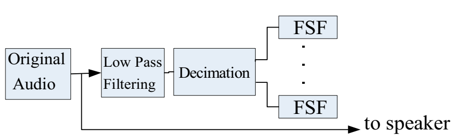


Fig. 3. Audio signal extraction block diagram

A. Engine RPM and the fundamental engine noise frequency

We set the sampling frequency to 1500Hz for the FSF filtering process. Consequently, the engine harmonic frequencies we picked up for analysis should be within the Nyquist frequency of 750Hz. For four stroke engine, the fundamental frequency is the product of the firing frequency and number of the cylinders, and the firing frequency is as follow

$$\textit{firing frequency} = \frac{1}{2} * \frac{\text{RPM}}{60} \text{ Hz.} \tag{3}$$

From the above equation, we have for four cylinder engine:

$$\frac{1}{2} * \frac{\text{RPM}}{60} * 4 \leq 750 \text{ Hz} \Rightarrow \text{RPM} \leq 22500 \text{ Hz}$$

22500 RPM is much higher than most consumer cars' engine maximum RPM, which is normally between 6000-10000 RPM. Depending on the engine noise profile, the engine noise harmonics selected can be different. When higher engine noise harmonics are selected, this range will be lower accordingly. For most consumer cars and with the objective of bass enhancement, the sampling frequency of 1.5 KHz and the RPM range is reasonable.

B. Parametric factor

For FSF filtering, there are two ways in settling the main parameters to control the filtering processing and centre frequency. One is to set the filter length N as a constant and change each of the frequency sample values. However, in this way, during the online reconstruction we need to change multiple sample values to achieve the reconstruction. On the other hand, if we first set the relative frequency samples at certain values, it is possible to achieve the reconstruction using only one parameter, the FSF filter length, N. For example, when we set H(k) for k=10 to be centered at the fundamental frequency, then we have:

$$\frac{Fs}{N} * 10 = \frac{1}{2} * \frac{\text{RPM}}{60} * 4 \Rightarrow \text{N} = [\frac{Fs * 10 * 30}{RPM}] \tag{4}$$

If RPM equals to 2500, the corresponding filter length would be 180. It's important to point out that, with advantages of FSF filter, longer of the filter length itself will not increase the computational load, as most frequency samples H (k) are zero, only the numbers of samples in passband will contribute to the computation.

C. *Transition band sample value*

Rabiner et al gave the suggested values for the values of the coefficients in transition bandwidth in [5]. Regarding the typical RPM from 1000 to 2500, we have our typical filter length from 180 to 450, and set three samples in passband, the optimum value for transition bandwidth sample would be around 0.4.

D. *Selecting suitable filter length/frequency resolution*

As the sampling frequency Fs is 1500 Hz, the frequency resolution for FSF filter would be $\frac{Fs}{N}$, where N is the filter length. According to the relationship

$$\frac{Fs}{N} * k = \frac{1}{2} * \frac{\text{RPM}}{60} * 4, \tag{5}$$

where k is the sample index we pick up to centre at the engine noise frequency. As a result, k will control the resolution of the filter. The optimal resolution is determined by the bandwidth of the engine noise harmonic. Off-line calibration is needed for different engine to select the proper k , which is to be set to the center frequency , correspondingly the frequency resolution.

3.2 Power estimation and post processing

In the signal power estimation, we use the exponential window method, which requires only one storage bin. The signal power estimation can be expressed as:

$$P_x(n) = \lambda P_x(n-1) + (1-\lambda)x^2(n), \qquad (6)$$

where $P_x(n)$ is the signal power, $x(n)$ is the current sample, and λ is known as the smoothing parameter or forgetting factor, typically set between 0.9 to 0.999 [6].

There are many options for the post processing block. Users can perform their desired processing, like different kinds of equalizing. In this paper, we use the equal-loudness compensation for the bass components to match the final audio to the human psychoacoustic perception [7].

Using the power estimation results and the compensators obtained, the gain factors β_i , $i = 1,2,..,n$ in the ANE systems, can be the calculated as:

$$\beta_i = \sqrt{P_i * C_i * \alpha}, \; i = 1,2,..Ns \qquad (7)$$

where P_i is the power of the FSF output at engine harmonic frequency, C_i is the equal-loudness contour compensator, and α is a constant to govern the sound volume in order to mix the tuned engine noise with the original audio output. Users can tune α to their own tastes, depending whether they want more bass or not. Ns is the number of the selected predominant engine noise harmonics.

3.3 Multi-frequency ANE system

To perform the final active control of the engine noise, we designed a multi-frequency consisting of several independent single-frequencies ANE systems, each tuning one harmonic frequency related to the engine noise. The number of the single-frequency ANE system needed is determined by the number of the selected predominant engine noise harmonics.

Each ANE block has its own gain factor self-tuned to the power of the related audio component. When the audio signal is changing with time, the equalization of the low frequency signal responds accordingly. The entire system diagram is shown in Figure 4.

4. EVALUATION BY RECORDED ENGINE NOISE

Performance of the proposed system was evaluated by recorded in-cabin engine noise using computer simulations. The engine noise was recorded in a Toyota Crown at passenger seat with the engine running at around 2600 RPM. The reference signal was generated by using cosine wave with the centre frequency at the engine noise harmonics. Kim et al. showed in [8] that the self generated reference good performance in ANC applications.

The audio signal used for evaluation is 'Hotel California' by The Eagles (live version). The clip was taken from the start of the track, which consists of a bass drum with slight audience noise. This makes it easier to focus on the bass.

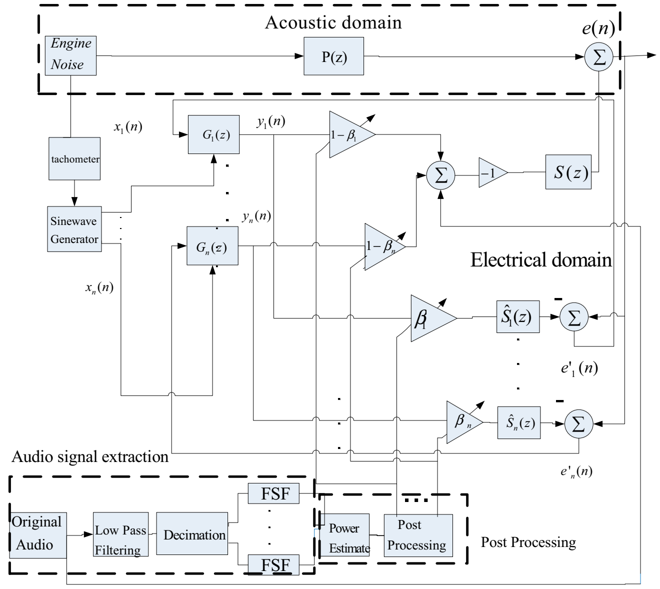


Fig. 4. System diagram

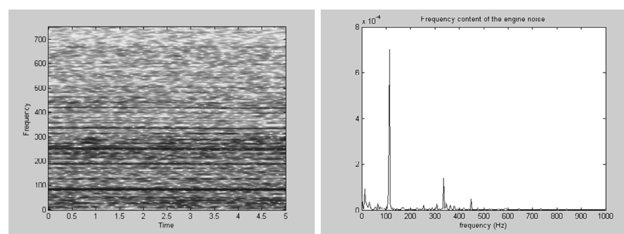


Fig.5. Spectrogram and power distribution of the noise

4.1 FSF filtering

Table 1 shows the offline calibration of the parameter k for the engine on Toyota Crown. From the table, we see the FSF extracts most of the signal power at the engine noise fundamental frequency when setting k =10 to the center frequency index. Correspondingly, the frequency resolution is 7.6Hz, and the passband width is double of the resolution of 15 Hz, as we use three samples in the passband.

Table 1: The setting of k for the center frequency

k	frequency resolution	FSF filter length	Power estimation
5	15Hz	98	25.2 %
8	9.5Hz	157	41.8%
9	8.5Hz	177	71.6%
10	7.6Hz	195	94.0%
11	6.9Hz	216	89.3%
12	6.3Hz	237	69.1%
15	5.1Hz	293	18.2%
20	3.8Hz	391	18.2%

Figure 6 shows the performance of the FSF filtering at engine noise fundamental frequency. The signal filtered is the engine noise sample, it can be clearly seen that the FSF extracts the frequency component centred at the engine fundamental frequency.

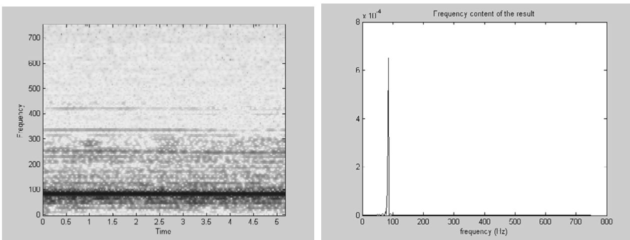
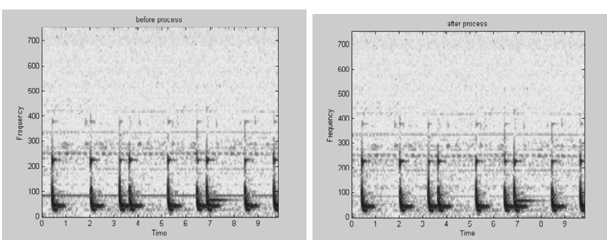


Fig.6. Spectrogram and power distribution of the filtered noise

4.2 Overall system

The results shown in Figure 7 are the bass components spectrogram of audio signal before and after the process. The predominant engine noise harmonics are cancelled when the audio is absent, and tuned according to the gain factor shown in Figure 8, when the audio is present.



(a) (b)

Fig. 7. Spectrogram of the sound in cabin when system off (a) and system on (b)

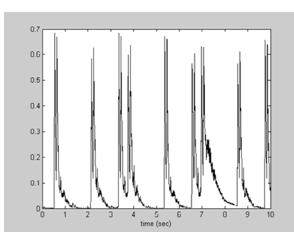
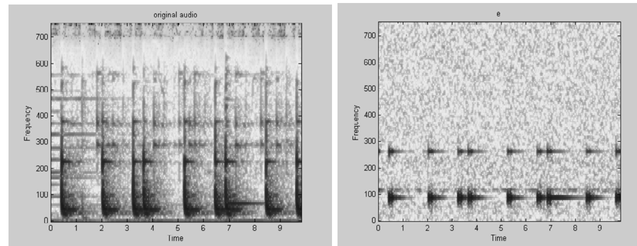


Fig. 8. Gain factor for fundamental frequency

To display the tuned engine noise more clearly, the spectrogram of the tuned engine noise alone is shown in Figure 9. It can be observed that the tuned engine noise has a similar spectrogram distribution with the audio signal. The residual engine noise components are used as the enhancement to the bass reproduction, while in ANC system all engine noise is supposed to be cancelled.



(a) (b)

Fig. 9. Spectrogram of the audio (a) and the tuned noise (b)

5. CONCLUSION

A novel method of enhancing bass audio reproduction in automobile cabin was proposed. Three sub-system of the method were introduced for achieving the enhance process. The FSF filter could be efficient in extracting audio components at certain frequencies and its recursive structure allows fast on-line reconstruction. We evaluated the proposed system with recorded real engine noise and audio signal. Simulation results showed that the proposed method presented could be a promising method of enhancing the audio bass reproduction in the cabin.

6. REFERENCES

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