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Real Time Suppression of Howling Noise in Public Address System

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Abstract

Howling noise is a common phenomenon in a public address system. It is built up due to the acoustic coupling between the speaker system and the microphone when it creates a positive feedback. Real time implementation of howling noise detection and suppression was implemented using TMS320C6713 DSK starter kit. The whole implementation was done based on direct memory access (DMA) feature of the DSP processor. The method uses the properties of howling noise for efficient detection and has the advantage of suppressing the noise. Howling detection is performed based on spectral flatness measure (SFM) of each input speech frame. For frames without howling, the input is passed as such to the output. Howling suppression is performed by making output samples as zero if the presence of howling noise is detected.

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Keywords: Howling noise ; Spectral Flatness Measure(SFM); Direct Memory Access(DMA).

1. Introduction

Public address systems are widely used for amplifying voice in various situations. It consists of a microphone, an amplifier and one or more speakers. Whenever we have those three components, there is a chance of feedback. Feedback occurs when the sound from the speakers makes it back into the microphone and is re-amplified and sent through the speakers again. This loop happens so quickly that it creates its own frequency, which we hear as a

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howling sound. The distance between the microphone and the speakers has a lot to do with the frequency of the howling, because that distance controls how quickly the sound can loop through the system. The acoustically closed loop exhibits a resonant frequency. The system automatically drifts into instability, when the loop gain approaches unity and an acoustic condition is created in the form of howling.

Reducing the gain or altering the position or microphone or speaker manually may alter the conditions required for sustained positive feedback and may suppress the noise. Instead of manual intervention, detection and suppression of howling noise in an automated fashion is very important in PA systems used in class rooms, seminar halls etc.The howling suppression systems try to remove the howling noise without distorting the speech signal. In early ages, the howling suppression was indeed by manual intervention as reducing the gain of the amplifier or changing the orientation of the microphone to alter the feedback condition. Later methods have developed which automatically detects and suppress the howling noise.

Adaptive periodic noise cancellation (APNC) [3] was a method proposed by J.B. Foley in 1989. It relies on the fact that howling is periodic, while speech is not when viewed over a long enough time scale. In this method howling noise is identified by an adaptive filter and then subtracted from the signal. The adaptive FIR filter is driven by the least mean square (LMS) algorithm. Using an optimum estimate of the primary correlation component of noise, the filter removes noise. To obtain good performance from APNC the bandwidth of the filter must be as narrow as possible.

Majority of the existing algorithms for howling noise suppression are designed using adaptive notch filter [4, 5,6]. In notch filter based howling noise suppression the gain is reduced in narrow frequency bands around frequencies at which the loop gain is close to unity. These methods require adaptive notch filters which instantaneously adjusts the filter coefficients to nullify the effect of howling frequencies. In this method of howling detection [4], the signal captured by the microphone is subjected to spectral and temporal analysis. Due to the sinusoidal nature of howling, the microphone signal frequency components having the largest magnitude are considered to be howling components. Spectral features for discriminating between the candidate howling component and tonal components is based on features like 1) PTPR: peak to threshold power ratio, 2) PHPR: peak to harmonic power ratio, 3) PAPR: peak to average power ratio, 4) PNPR: peak to neighboring power ratio, 5) IPMP: inter frame peak magnitude persistence, and 6) IMSD: inter frame magnitude slope deviation. The temporal features for howling detection rely mainly on i) the persistence of howling components for a longer time, ii) exponentially increasing amplitude until the saturation. A high probability of false alarm results in poor sound quality due to the unnecessary activation of notch filters, a low probability of detection may also lead to poor sound quality because of howling. The logical conjunction of single feature criteria decreases the probability of false alarm but has no effect in the probability of detection. Three novel criteria has been defined by combining pair wise logical conjunction of PHPR, PNPR, and IMSD criteria and an additional three-feature criterion results from combining all three PHPR, PNPR, and IMSD features. The novel criteria exhibit a drastic decrease of the false alarm for a fixed probability of detection. The detected howling frequency is removed by activating corresponding notch filter from a bank of adjustable notch filters.

Howling detection based on LMS adaptive notch filter and phase locked loop [5] has very low computational complexity and short detection delay. The difference of this method is in the use of a phase locked loop for the realization of the 90-degree phase shifter. Here howling detection is achieved by adaptive notch filter with only two adaptive parameters.

In a DSP-based acoustic feedback canceller for public address systems [2], howling detection is accomplished by performing three tests on the strongest peak frequency. Initially the peak power component is checked against a threshold above which it is identified as possible acoustic feedback. The second test determines the average power of last N samples which establishes a relative threshold against the peak frequency can be checked. The third one is the test for the second harmonic power content. It is based on the fact that speech has strong harmonic components while feedback noise has less or no harmonic components. A notch filter with notch frequency coinciding with the tone is used for removing the feedback signal. A two tap IIR filter has selected over a 25 tap FIR filter because of the ease of implementation and the minimal storage space. The filter coefficients are pre-calculated and stored in memory. Characteristics of each filter can be determined and fine-tuned beforehand for optimum performance. To further improve the performance a gain reduction technique is used which reduce the system gain when howling is continuously detected. This method does not audibly distort the transmitted signal. The action of the notch filter attenuating with a bandwidth approximately equal to frequency resolution obtained by the FFT used, is hardly audible. This method may not detect clicking sounds that may occur occasionally. In such cases, several frequencies may fulfil the feedback criteria needed for howling.

Adaptive howling suppression (AHS) in an audio amplifier system [6] has remarkable advantages over the conventional adaptive noise cancellation using adaptive filters. The disadvantages such as excessive coefficients requirement in adaptive feedback cancellation and slow adaptation of LMS algorithm are eliminated in this method. The AHS method requires less number of coefficients of the system and performs as a notch filter characteristic for the howling spectrum reduction. The variable momentum least mean square (VMLMS) algorithm is proposed to combat the rise in mean square error with convergence rate in LMS and NLMS algorithms. VMLMS algorithm has high convergence rate with low excess MSE. The AHS structure is composed of an adaptive FIR filter and a non adaptive filter. Each update of the adaptive filter coefficients are copied to the non adaptive filter which is then worked together with feedback constant to provide an IIR form.

Howling suppression can be performed using FFT and chirp Z transform. A wave with reversed phase and amplitude and frequency equal to that of the howling signal is used to suppress the howling [7]. Statistical analysis of temporal spectra is used in [8] for howling suppression.The temporal variation in the power spectra is random at stable conditions whereas assumes small values at a specific frequency in howling condition. The standard deviation of power spectra is used for the detection of howling which has a noticeable difference in values between stable and howling conditions. The algorithm consists of calculating temporal variation in the power spectra and peak frequency simultaneously. The standard deviation is calculated from the moving averages of power spectra. A suitable threshold is used to detect the howling. The complexity of this method is very low compared with the conventional LMS algorithm.

In this paper we propose a method for automated detection and suppression of acoustic howling based on spectral flatness measure (SFM).

2. Method for Howling Noise Suppression Based On SFM

The block diagram representation of the howling detection and suppression system is shown in Fig. 1. The input speech from microphone is fed into a howling detection stage and to a block of gain. The presence of howling is detected by the howling detection stage. The gain of the signal is appropriately adjusted by the gain selection logic. Once howling is detected, gain is zero and zero output samples will be sent out. If there is no howling, input is passed as such to the output.

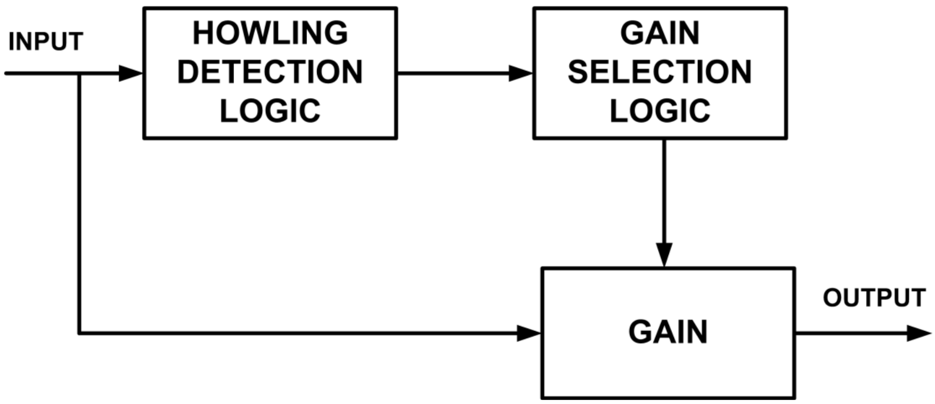

Fig. 1. Block diagram of howling noise suppression system

2.1. Howling Detection Logic

A measure of spectral flatness is used to measure the flatness of magnitude spectrum of the signal and to check the presence of howling noise [1]. The sampling frequency used is denoted by f_s. Hamming windowed frames with 50% overlap is used for signal processing. The magnitude spectrum $X(n,k)$ is computed using 1024 point FFT. This gives frequency resolution adequate for processing. Here n denotes the time index and k is the frequency index. We have used 20 spectral bands in the frequency range 0 to $f_s/2$, each band hawing a bandwidth of $f_s/40$ Hz. For frame number n, we compute the spectral flatness measure (SFM) for spectral band b using

$$\text{SFM}\ (n,\ b) = \frac{\left(\prod_{k=bl}^{bu} X(n,k)\right)^{1/w}}{(1/\mathrm{w})[\ \sum_{k=bl}^{bu} X(n,k)\ ]} \tag{1}$$

where b_l and b_u are the frequency indices for the lower and upper boundaries of band b, and w is the width of the band. Spectral flatness measure is computed as the ratio of geometric mean to the arithmetic mean. It approaches 0 for a peaky spectrum. It has a value of 1 for a flat spectrum. When the howling noise dominates, the SFM approaches 0 in the corresponding frames. Howling in a band b is marked by a flag updated as

$$H(n,b) = \begin{cases} 1, & if\ \text{SFM}(n,b) < \theta_h \\ 0, & otherwise \end{cases} \tag{2}$$

where θ_h denotes the threshold used for detection of howling.

An evaluation was performed for the detection logic based on SFM using a computer based setup for pre-recorded speech. Different howling noises were recorded by varying the amplifier gain. It was recorded at sampling rate of 16 kHz. Each waveform is processed with Hamming window of duration 30 ms and SFM values are computed for all frames. Frames with howling are detected using these SFM values and the frequency of howling noise is estimated.

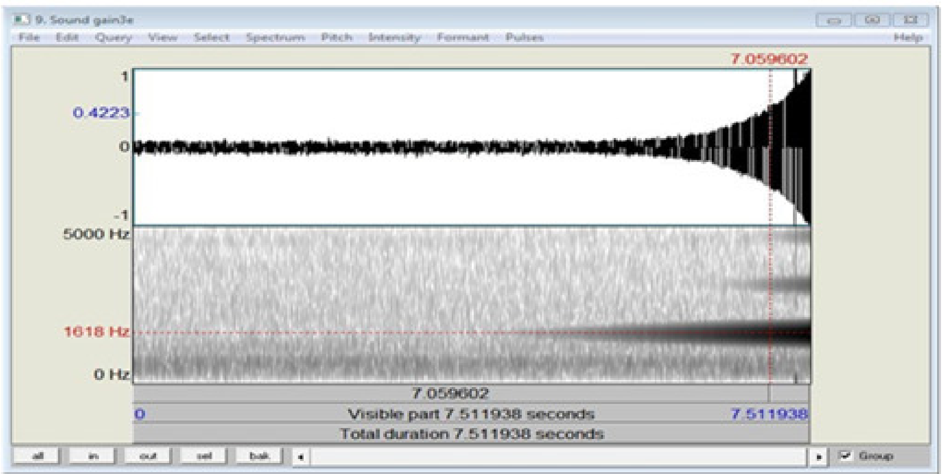

Fig. 2. Howling noise waveform and its spectrogram

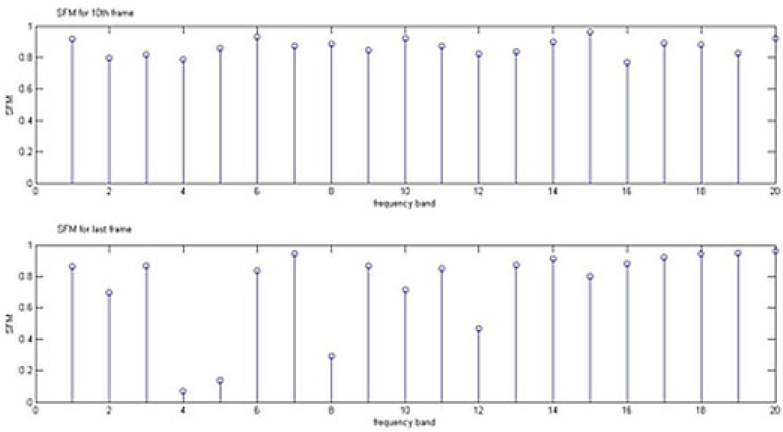

Fig. 3. SFM values for the example waveform given in Fig. 2.

One example waveform of howling noise recorded is shown in the Fig. 2. Upper part of the figure represents the time domain behaviour and lower part indicates spectrogram of corresponding waveform. In first few seconds there is no howling. Howling starts to build at a particular position of microphone and speaker for a particular gain setting and room condition and it builds up to saturation. From the spectrogram it can be seen that howling frequency is around 1600 Hz.

For the above waveform, SFM values for frame with howling and without howling are shown in Fig 3. Initially there is no howling and SFM values of the corresponding frame (10^{th} frame is chosen here) is near to 1. Last frame is selected to check the presence of howling and SFM value of the 4th band in that frame is near to zero, which corresponds to a frequency 1600Hz.

2.2. Gain Selection Logic

The gain of the signal path is adjusted by the gain selection logic. In the presence of howling, the gain is set to zero and the output frame is with all zero valued samples. In the absence of howling, gain continues as unity thereby giving samples in the output frame same as that of the input frame.

3. Real Time Implementation of the Algorithm

Real time implementation of the howling noise suppression algorithm utilizes the Enhanced DMA capabilities of the DSP processor TMS320C6713. Direct Memory Access (DMA) transfers data between memory and peripheral locations without the intervention of the CPU. The data transfers take place in parallel with CPU activity, maximizing system performance. Data can be transferred concurrently with CPU transactions. Since the data transfers are not interrupt driven, the system performance is maximized.

We have configured two DMA channels to implement the algorithm for howling noise suppression. One channel is from McBSP DRR (Data Receive Register) to internal memory and other channel is from internal memory to McBSP DXR (Data Transmit Register). The EDMA is configured to take every 16-bit signed audio sample arriving on McBSP1 and store it in a buffer in memory until it can be processed. Once it has been processed, the EDMA controller sends the data back to McBSP1 for transmission. Two buffers are used for both transmission and reception. So there are total 4 buffers used for proper EDMA operation. When an EDMA transfer complete, it will generate an interrupt and processing of the received data will start. While the 1st buffer is being filled, the second buffer can be processed with the knowledge that the current EDMA transfer won't overwrite it. For completion of 1 EDMA transfer it will take a time interval equal to the buffer size multiplied by 1 sample period. The signal processing required for howling detection and suppression can be performed within this time duration. Howling detection was performed based on spectral flatness measure (SFM). Sampling frequency of the CODEC was set at 8 kHz and SFM was calculated for each frame. Since the frame length was taken as 300 samples, EDMA buffer size was configured as 300. Processing of these 300 samples can be performed while the other buffer is being filled. Presence of howling is detected based on a flag *H*. If any of the 20 bands in a frame has an SFM value less than the threshold, the flag will be set. Then that frame is suppressed and zero output samples will be sent out to the speaker. Optimum value of the threshold found is 0.15. By default, *H* is in reset condition and same input signal is sent to output as such.

4. Real-time experimental Results

Fig.4 shows the values of the received buffer when input signal is pure speech signal without any howling noise. Fig.5 shows the values of the received buffer when the input signal is contaminated with howling noise. The waveform in Fig.4 is a sine wave with a particular frequency which demonstrates the tonal property of howling noise.

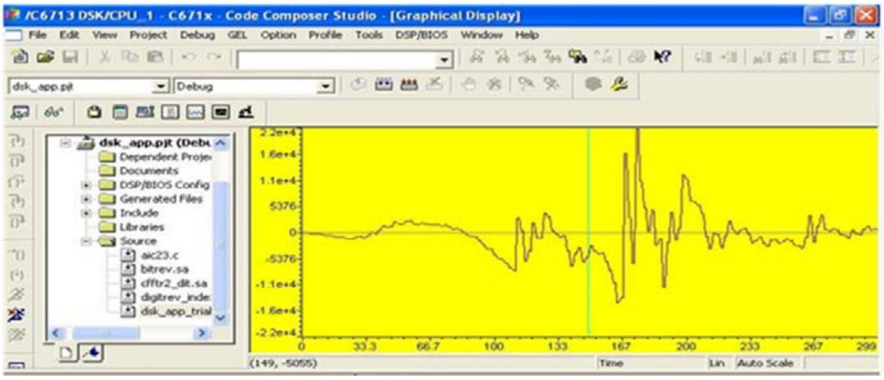

Fig. 4. Speech waveform in the absence of howling.

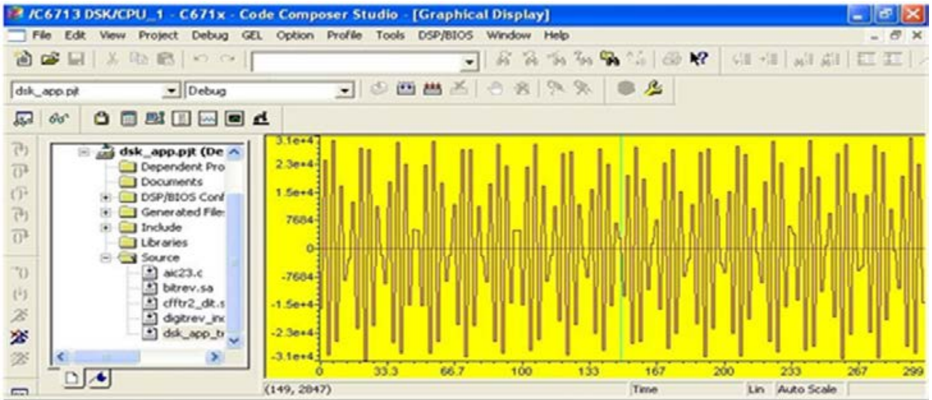

Fig. 5. Howling noise waveform

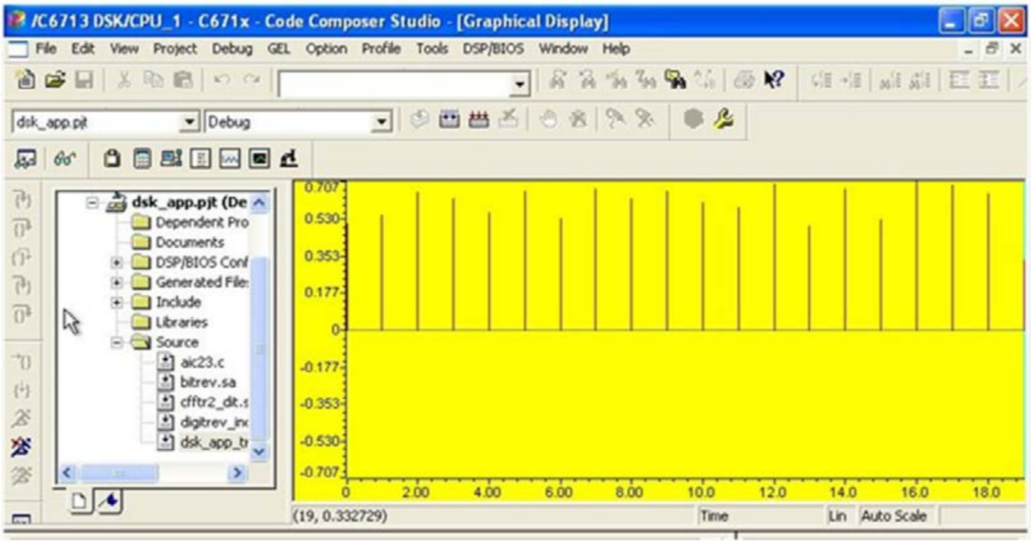

Fig. 6. SFM values of the speech waveform

Values of the SFM buffer are shown in Fig.6 and Fig.7. SFM values of the speech signal frame is shown in Fig. 6. It can be seen that SFM values are near to 1. When howling occurs, some of the bands in the frame have SFM values near to zero and this is shown in Fig.7. From the figure it is observed that less SFM values are in 8th, 9th and 10th bands and values are 0.094, 0.084 and 0.124 respectively. Since sampling frequency used in real-time processing is 8 kHz, each band has a width of 200 Hz. So howling frequency is around 1800 Hz.

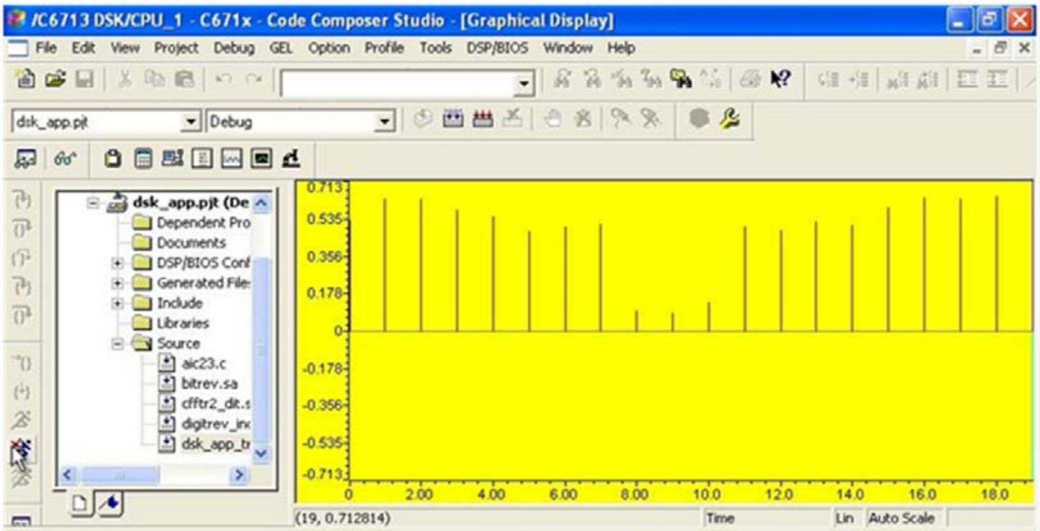

Fig. 7. SFM values of howling noise waveform.

5. Conclusion

Howling is very annoying to the speaker and audience and suppression of howling noise by automated means is of high practical importance. This paper was aimed to develop a real-time howling suppression system which can be used in class rooms, seminar halls etc, where speech from a single speaker is captured by a single microphone. An offline implementation of the howling detection algorithm was implemented in MATLAB and was tested using pre-recorded speech. Real-time implementation of the howling noise detection and suppression system was carried out using TMS320C6713 DSK. The whole implementation was done based on direct memory access (DMA) feature of the DSP processor. The algorithm based on DMA for sample acquisition and buffering is more efficient than the algorithm running on interrupts. Howling detection is done based on spectral flatness measure (SFM) of each input speech frame. For frames without howling, the input is passed as such to the output. Howling suppression is performed by making output frames with zero valued samples, once the presence of howling noise is detected. Further modifications are required in the howling detection part of the system such as the possibility of using speaker adaptive and dynamically updated thresholds for SFM. The effect of the howling suppression system for non-speech input signals from musical instruments etc. needs to be investigated.

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