PERFORMANCE ANALYSIS OF AUDIO SIGNAL DETECTION AND JAMMING

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Abstract— *In this paper, system on a, analyzing of audio signal detection and jamming, this can be used to jamming the sounds inside the building. In general, audio signal is jammed by giving back the frequencies to incoming frequencies. This effect will avoid disturbance to the patients without any physical discomfort. We utilize this phenomenon and implemented by noise detection and band pass filtering, where the frequency is generated with the help of Tone generation, disabling the sound signals inside the hospitals. We discuss performance analysis scenarios of the system, such as facilitating and jamming discussions. Finally, the system parameters are analyzed and the audio signal jamming is done. Moreover, by this process the signals can be jammed below the 4500Hz.so that it is safe for the patients in the hospitals.*

Key words— *Audio signal jamming, Tone generation, Sound signaling*

1.INTRODUCTION

Sound noise is a promising problem in the world. Sound noise causes hyper tensions to the patients in the hospitals. Sounds are of many categories such as sounds of vehicles on the road, sounds of crackers, drilling machines, thunder etc. The human ear can be to audible frequency range, roughly 20Hz-20kHz. The normal human ear can detect the difference between 440Hz and 441Hz.since the hospitals are constructed near the roadside areas it disturbs the patients frequencies between 2,000 and 5,000 Hz. So, to overcome these problems, the proposed system is sound jammers. The sound jammers are mainly designed to block the outside unwanted sound into the building .The system is designed in the form of a shield attack jammers. The jammer is an EMF shielding. This is done in a closing area in a faraday cage so that the person inside the building cannot receive audio signals from outside of building .This jammer does not affect the human being since it jamms only the frequency of sound upto a certain range . In this paper, audio signal is received by the receiver where, The frequency of any unknown audio signal can be calculated using Fast Fourier Transform (FFT). FFT is performed on the acquired audio data to find the frequency of the audio signal. The unknown frequency of has been determined. The maximum human frequency range is 2,000 and 5,000 Hz. Some pitch sharpening mechanism must be operating. Since the ear is surrounded by air, the sound waves are constrained.Normal range of sound pressure and sound intensity may also be specified

2. METHODOLOGY

Initially the sound of the vehicles on the road , sounds of the crackers are received by the noise detector. The information from noise detectors is then passed through the frequency meters .The frequency meters determines the frequency of the received noise from the noise detector. Then only certain range of frequency is passed through the band pass filter. Then those frequencies are processed by the algorithm. Then the frequencies are added. By frequency combining the jamming can be done. The jamming is done in the shielding attack

3 . NOISE DETECTION

Initially the sound of the vehicles on the road, sounds of the crackers are received by the noise detector. The information from noise detectors is then passed through the frequency meters . To estimate the DFT of N points in the naive way,by using the definition, it takes O(N2) arithmetical operations, while an FFT can estimate the same DFT in only O(N log N) operations. The difference in the speed can be enormous, especially for real time data such as sound wave or speech signal where N may be in the thousands .For 1024 samples a straight DFT requires 10242 = 1048576 arithmetic operations. However the same number of samples the FFT requires 1024*log2 (1024) = 10240 arithmetic operations. The frequency meters determines the frequency of the received noise from the noise detector. Then only certain range of frequency is passed through the band pass filter. Then those frequencies are processed by the algorithm. SOUND noise detectors is typically used in detecting the loudness in ambient. Sound is detected in the form of analog waveform Sound of the vehicles on the road , thunder, sounds of the crackers are received by the noise detector detects the natural sounds. The sound is detected and processed

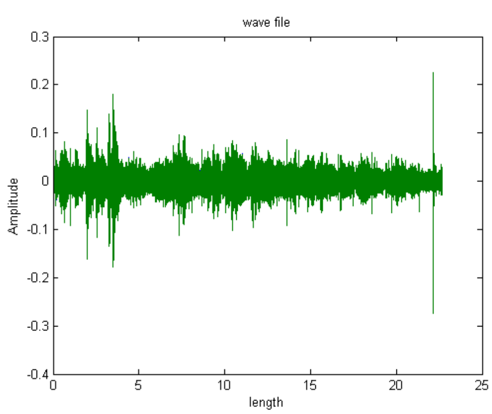

Fig 3.1 Audio waveform of the vehicle sound

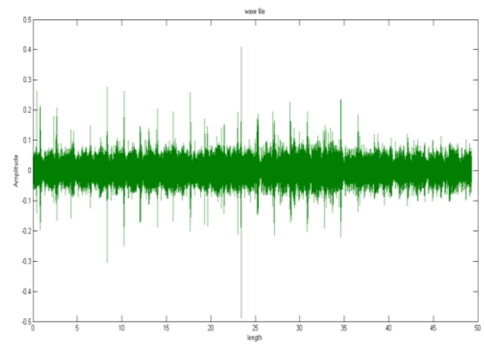

Fig 3.2 Audio waveform of the orchestra

4 FAST FOURIER TRANSFORM

Fast Fourier Transform is an algorithm that, this is mainly to determine the frequency of the audio signal. Fourier analysis is done with the audio signal from its original to a representation to calculate the frequency of the audio signal. A FFT computes such type of conversion by factorizing the DFT matrix into a product of inadequate factors. As a result, it manages to minimize the complications of computing the DFT from O(n2).An FFT is a way to compute frequency of the audio signal more with less complexity. This mainly computes the "N" DFT points, this counts O(N2) arithmetic operations, while FFT can compute the same DFT in O(N log N) operations. The difference in the speed can be high, mainly for real time data sets where ,N may be in the thousands or millions values. In practice, the computation can be reduced by several orders of magnitude, and the improvement is approximately proportional to N / log(N) values. This improvement makes calculation of the DFT efficient practical; FFTs are used in wide variety of applications, in digital signal processing and and audio signal processing.

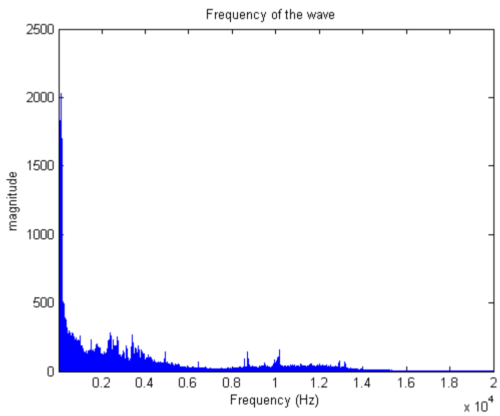

Fig 4.1 frequency of vehicle sound

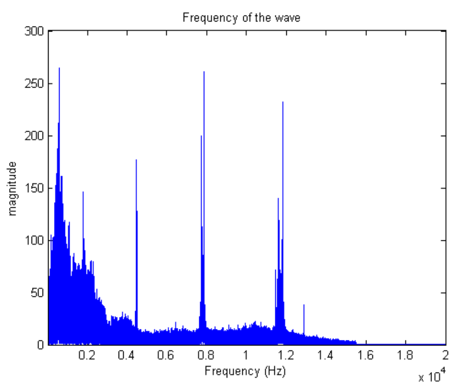

Fig 4.2 frequency of Orchestra music

Several types of sound were collected such as sounds of vehicles on the road. The frequency of the audio signals are calculated.

5. BANDPASS FILTER

The band pass filter in audio processing mainly to stop the frequency of the audio signal to a certain range. This band pass filter is used here only to pass frequencies below 4000Hz.The frequencies above the 4000Hz are stopped since it disturbs the patients in the hospitals. One simple use for these types of Passive Filters is in audio processing applications or circuits such as in filters or pre-frequency tone controls. This is done by connecting" together a Low Pass Filter circuit with a High Pass Filter circuit, so that we can limit the frequencies we can with passive RC filter that passes only selected "band" of frequencies. This new type of passive filter layout produces a frequency selective type of filter known commonly as a Band Pass Filter or BPF for short. This process is mainly done at the receiver. This process carried out Inverse impulse response with Chebyshev type I technique. The cut-off frequency point is calculated by a simple RC passive filter which controls using just a single resistor in series with a non-polarized capacitor, and depends on which ,way they are connected that can be either narrow or wide while weakening all those outside of this range. This bandpass filter while designing with chebshev type I technique stops the audio signals below the 4000Hz frequency range

6. INVERSE IMPULSE RESPONSE

Inverse Impulse Response is a technique applying to many linear time invariant Systems . The inverse impulse response is mainly used in the receiver.the inverse impulse response uses many types of filters such as Chebyshev type 1,chebyshev type II , butterworth filter. This impulse response does not become exactly zero at a certain point, but continues indefinitely. This is in variance to a finite impulse response in which the impulse response $h(t)$ *does* become zero at a time $t > T$ for some finite T, for being in finite duration.

The main advantage of IIR filters have over FIR filters is their efficiency in implementation, pass band, stop band , ripple and/or roll-off factor.

7. CHEBYSHEV FILTER

Chebyshev type I band pass filters usually derives filtering polynomials from general Chebyshev filtering function are mainly to filter the audio signals. This have a steeper roll-off and gives exact filtering than Butterworth filters. Chebyshev type I filters have the property that they reduce the error between the idealized and the definite characteristic over the range of the filter, but ripples in the pass band. This because they are expressed and it is derived from Chebyshev polynomials.

$$G_n(\omega) = |H_n(j\omega)| = \frac{1}{\sqrt{1+\varepsilon^2 T_n^2\left(\frac{\omega}{\omega_0}\right)}} \quad \text{........(1)}$$

where ε is the ripple factor. ωo is the cut off frequency and Tn is a chebyshev polynomial of the order n and thus ,The passband exhibits equiripple behavior, with rippling determined by the ripple factor ε. In the pass band, the Chebyshev polynomial value falls between -1 and 1 so the filter gain is maxima at G = 1 and minima at

$$G=1/\sqrt{(1+\varepsilon^2)}. \quad \text{...... (2)}$$

At the cutoff frequency ωo the gain again has the value

$$1/\sqrt{(1+\varepsilon)}) \quad \text{.......(3)}$$

but continues to reduce the stop band as the frequency increases. The common practice of determining the cutoff frequency at −3 dB is usually not applied to Chebyshev filters; on preferably the cutoff is taken as the point at which the gain falls to the value of the ripple for the concluding time.

The order of Chebyshev filter is exactly equal to number of active components needed to estimate the filter .

The ripple is often given in

$$\text{dB:}20\ \log\sqrt{1}+\varepsilon^2 \quad \text{.........(4)}$$

So that a ripple amplitude of 3dB results from $\epsilon = 1$

An even steeper roll-off can be obtained when ripple is allowed in the stop band .by allowing zero on the jω-axis in the complex plane.however ,this results in less eradication in the stopband.the result is also called as cauer filter.

The given below figure is the designed bandpass filter with respect to the sampling frequency of 24000Hz and the frequency is limited to 4000Hz

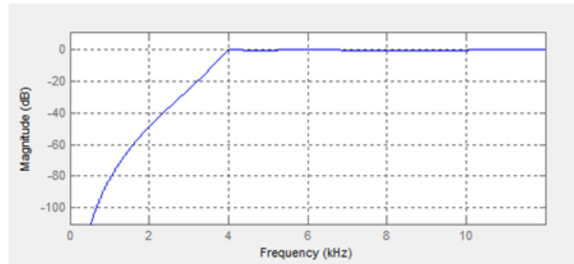

Fig 7.1 Designed chebyshev filters

The waveform is then the frequency of the signal is limited to 4000Hz.

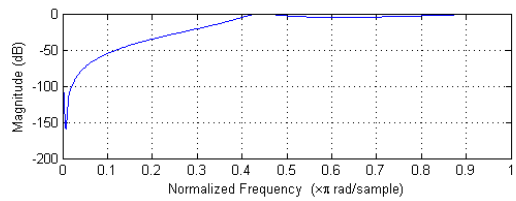

Fig 7.2 Band limited frequency of Orchestra music and vehicles sound

8. AUDIO SIGNAL GENERATION

Audio signal generator is mainly used to generate frequency. This audio signal generation mainly generates frequency with the help of Tone generator. This tone generator is mainly used to generate audio frequency. This toned frequency is then subtracted with the band pass filtered signal of 5000Hz. Thus by subtracting the audio frequency is jammed..

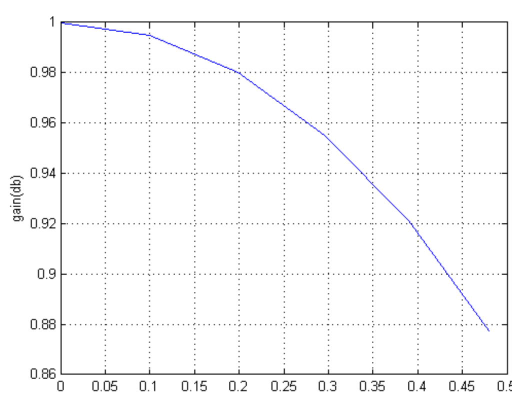

Fig 8.1 Analysis of jammed audio signal

9. CONCLUSION

In this paper that a system is designed to pass the signal below 4000Hz in the building. This can be reached, even under important noise climatic conditions. The study is presently undergoing on, taking more complex and real noise environment conditions into account. Hybrid solutions seem to be compelling in order to increase robustness and reduce the system complexity load. At system level, detection and jamming of signals above this 4500hz will be done and analyzed.

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Biographies

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