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Effects on Internet Signal Processing

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ABSTRACT
Internet signal processing is the means of processing internet modulated or demodulated data in order to determine the content of the data. In modulation of internet signals, sometimes it is difficult to receive the actual transmitted signal due to loss of signals which is known as attenuations or can be hindered by an external or internal interference which may lead to noise causing distortion to the signal. In this work the internet audio signal wasconsideredand how interference cause an effect on the audio internet signal, this effect may lead to cracking of sound signal, noise or echoes. The method used in analyzing this spectrum is the discrete time Fourier transform and with MATLAB/SIMULINK,where system audio signal was simulated to observe the unfiltered spectrum with noise that causes attenuations in the channel and also filtered with a finite impulse response in reducing the noise in the channel.

KEYWORDS: Signal, Processing, Internet, Simulink, FILTER, FIR

I. INTRODUCTION

Signals are processed for a variety of purposes, including the removal of unwanted noise, the correction of distortion, the preparation of the signal for transmission, and the extraction of specific meaningful information. The mathematical underpinnings of analog and digital signal processing will be provided in this work. There will also be some introductions of actual hardware implementation examples. Analog signals are produced or used by the vast majority of process control and audio equipment in the field, including sound sensors, transducers, and control elements.

However, it is highly likely that digital technology underlies the entire procedure. As a result, a variety of signal-processing techniques must be taken into account in applications for instrumentation and process control [7]. Despite the fact that signal processing is a very broad topic, we will only consider one aspect of it, such as internet sound signal processing. In the transmission of signals over long baselines, the implementation of compensating time delays, and the measurement of cross-correlation of signals, the use of digital instrumentation offers significant practical advantages over analog instrumentation.Long delays accurate to tens of picoseconds are easier to achieve digitally than by using analog delay lines, where the accuracy of the delay depends on the precision of the timing pulses in the system. Furthermore, other than the quantization effects that can be calculated, the signal is not distorted by the digital units. In contrast, when delay elements are switched in and out of the signal channels in an analog system, it can be challenging to maintain the shape of the frequency response within tolerances [7]. As required for spectral line observations, digital correlations, including those with multichannel output, are easily implemented.

Filter banks are used in analog multichannel correlators to separate the signal passband into numerous narrow channels. When exposed to temperature changes, these filters may cause phase instability. Finally, digital circuits require less adjustment than analog ones and are better suited to replication in large numbers for large arrays, with the exception of the highest bit rates (frequencies). The voltages must be quantized so that each sampled value can be represented by a finite number of bits before the signal waveforms can be digitally encoded. Typically, there aren't many bits in each sample, especially when the signal bandwidth is wide and high sampling rates are needed [6].

The modification of the signal levels to the quantized values effectively adds a component of "quantization noise," so coarse quantization results in a loss in sensitivity. This loss is typically negligible and outweighed by the other benefits. Sensitivity and complexity must be traded off when designing digital correlators, and the choice of how many quantization levels to employ is crucial [1]. A random noise signal can have its spectrum determined in one of two ways. You can measure the signal's autocorrelation function and then perform a Fourier analysis. after a predetermined amount of integration time, transformed into a power spectrum [3].

Alternatively, the signal could first undergo Fourier transformation before the square modulus is obtained. In the first scenario, the autocorrelation function's calculated autocorrelation function's resolution is roughly the reciprocal of the number of lags. The data stream must be segmented in the direct Fourier transform route in order to control the spectral resolution, which is roughly equal to the length of the data segments. Over the course of the integration period, the power spectra from every segment are added. The number of lags in the correlator is set equal to the number of segment samples in order to compare the results of these methods. The same two techniques can be utilized for interferometry [2].

The discrete time Fourier transform will be used in this research project to analyze audio internet signal processing.

When interference occurs on the channel used to transmit the signals, it affects internet signal processing by introducing noise or distorting the signals produced by digital signal processing (DSP) techniques, primarily digital filters. Image synthesis, video processing, or sound processing are all involved in this. In order to create new modified sounds with desirable effects for specific applications, sound processing uses one or a small number of prerecorded sound clips that go through one or more stages of modification through processing or filtering [8].

II. METHOD

Investigation of Effect of Internet Signal Processing

Investigating the impact of internet signal processing from the ground up using software and hardware. This feature can produce electronic music, interactive performances in video games, ring and alert tones for mobile cellular devices, among other things, through the internet. Particularly intriguing is the synthesis of internet signal processing's effects [4]. We present a model for the impact of internet synthesis based on how various materials react to attenuation. When considering how the internet affects sound synthesis, it must be based on the opposite, or sound analysis. Finding the individual frequencies in a sound's spectrum is all that is required to analyze it. Then, by altering the amplitude, phase, and frequency of the corresponding trigonometric components, it is possible to produce various sounds [6].

When you take into account, for example, the constrained audio memory budgets of gaming consoles, the benefit of sound synthesis becomes obvious. Pre-recorded clips offer excellent quality and do not require computation, but they must be stored in memory because internet streaming has an excessively high latency. As a result, sound synthesis is seen as a desirable approach to the sound issue [6].So, we can now express the sound signal retrieved from the internet using Discrete time Fourier Transform as [6]

$$x(\omega) = F(x[n]) = \sum_{n=-\infty}^{\infty} x[n]e^{-j\omega n}$$
(1)

Where the internet sound signal can also be expressed by it inverse Fourier Transform

$$x[n] = F^{-1}\big(x(\omega)\big) = \frac{1}{2\pi}\int_0^{2\pi} x(\omega)e^{j\omega n}d\omega$$
(2)

$$x[n] = e^{j\omega 0n} \quad (3)$$

3.2 Internet Signal noise filter with FIR Low Pass Filter.

FIR Filter Design by Windowing In designing FIR filter, given the frequency response Hd(ejω) and impulse response hd[n] of an ideal system, we would like to approximate the infinitely long hd[n] with a finite sequence h[n], where h[n] = 0 except for $0 \le n \le M$. Consider an ideal low pass filter whose frequency response is finite and rectangular [6]. A possible approximation error criterion can be defined as

$$E = \frac{1}{2\pi}\int_{-\pi}^{\pi} |H_d(e^{j\omega}) - (e^{j\omega}|^2 d\omega$$
(4)

To minimize Error, use Parseval's theorem:

$$E = \frac{1}{2\pi}\int_{-\pi}^{\pi} |H_d(e^{j\omega}) - (e^{j\omega}|^2 d\omega$$
(5)

$$= \sum_{n=-\infty}^{\infty} |h_d[n] - h[n]|^2$$

$$= \sum_{n=0}^{M} |h_d[n] - h[n]|^2 + \sum_{n=z[0,M]} |h_d[n] - h[n]|^2$$

$$h[n] = \begin{cases} H_d[n] & 0 \le n \le M \\ 0 & \text{otherwise} \end{cases} \quad (6)$$

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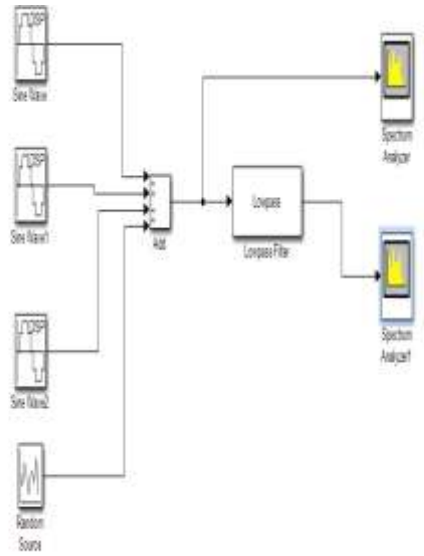

Figure 1: internet signal simulations diagram using Simulink

Internet Sound signal Synthesis Model

$x(t) = \sum_{m}^{M} g_{m\ Am}(t)\sin(2\pi f_m(t)+\emptyset_{m+r(t)})$
(7)

The model for internet sound signal synthesis can be described by the following equation which combines the different variables that can be adjusted to produce different impact sounds.

where $A_{m(t)}$ is the possibly time-varying amplitude, $f_m(t)$ is the possibly time-varying frequency, $\emptyset_m$ is the phase and g_m is a constant, all related to the m-th sinusoidal component, and finally r(t) is a noise component [5] [6].

$A_m(t)e^{-at}$ α is a decay constant.
(8)

x(t) will them be a sound of impact (when an object is struck for example). We claim that the impact sound is produced through additive synthesis. The term "additive synthesis" refers to a group of related synthesis methods that are all predicated on the notion that more complex tones can be produced by combining or adding simpler ones. Any complex sound can be divided into a variety of simpler tones, typically in the form of sine waves (Fourier) [6].

III. RESULTS AND DISCUSSIONS

Unfiltered Signal

When an internet signal is being transmitted through a channel, the digital signal is expected to get to the receiver by a demodulated processed. But along the line sometimes the signal will get interfered by an external or internal intrusion, thereby the effect of this interference will now lead to attenuations, or noise which will now be a negative effect on the signal about to be processed. From the diagram in figure 4.1 shows a distorted internet signal by a Gaussian noise, thereby reducing the internet sound quality.

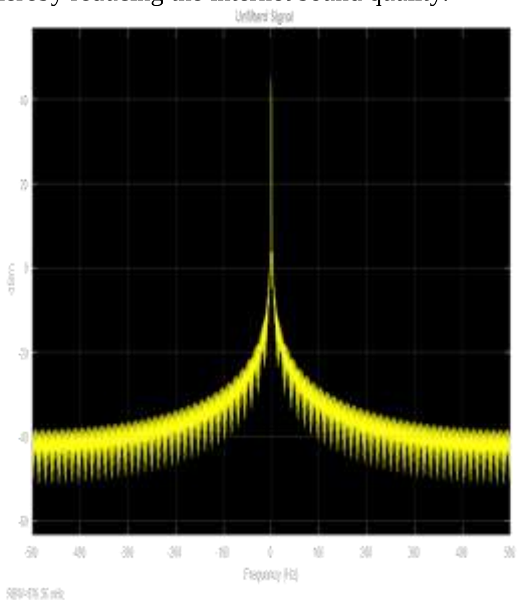

Figure 2: Unfiltered Signal

Filtered Noise in Internet signal

The major issues in retrieving the internet signal are distortions due to noise and other forms that may lead to attenuation thereby hindering the receiver to demodulate the actual signals. From the figure 4.2 describes the filtered internet sound signal with an improved quality.

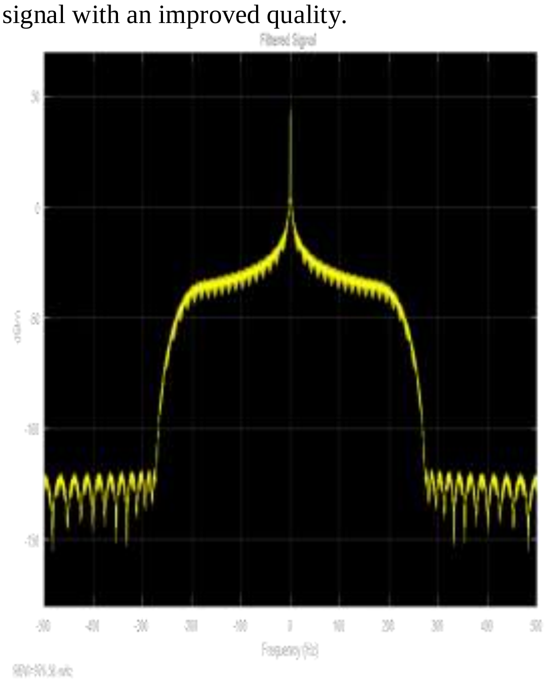

Figure 3: filtered signal

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Figure 4.3 show an internet signal with some noise signal preventing the receiver to get the actual signals.

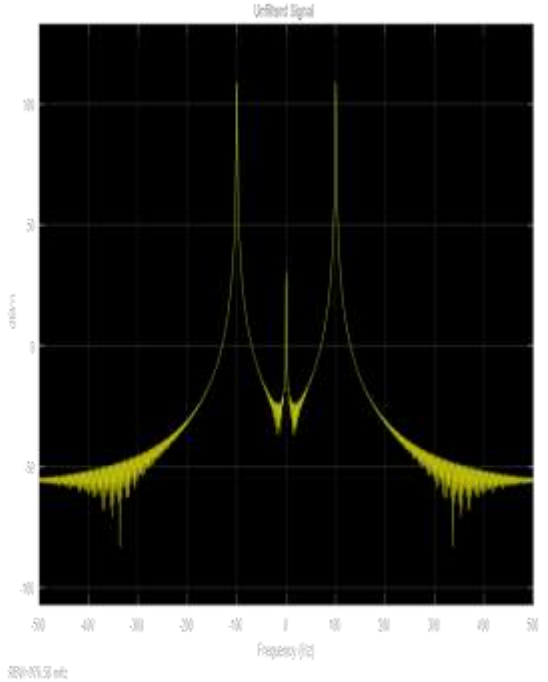

Figure 4: Unfiltered noise in the spectrum

Figure 4.4 describes the filtered noise from the internet signal thereby maintaining a smooth signal for the receiver thereby reducing the noise effect in the spectrum and improving the signal strength.

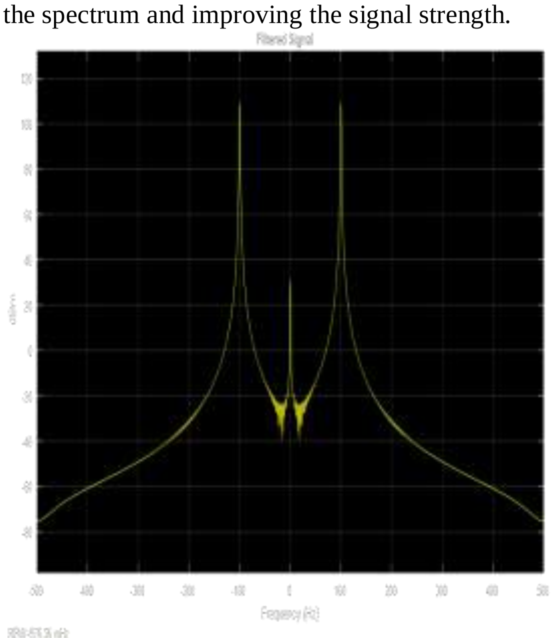

Figure 5: Filtered spectrum with a smooth shape of reduced noise in the spectrum

IV. CONCLUSION

Effect on internet signal processing is a novel approach on observing the several effects in the data spectrum. However, we have kindly taken into consideration of the internet audio signal thereby observing the effects on audio internet signal using Discrete time Fourier transform. Further approach was taken to observe the effects by carrying out a Simulink simulation, which we were able to observe the distortions and also filtered the noise in the transmitted signal which we were able to achieve an improved signal in the spectrum with a reduced noise. Finally, we can say that this method used in reducing the noise in the spectrum is an effective and a better method which comprises of a DSP sine wave, Gaussian noise and a finite impulse response low pass filter.

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