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Mathematical Modeling For High Performance Communication Using Next Generation Light Fidelity(Li-Fi) Technology

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Abstract
Light Fidelity (Li- Fi) is a wireless optical networking system that transmits data using light-emitting diodes (LEDs). This technology results in high speed, bi-directional wireless communication with the help of optical sources and detectors. This work focuses on the mathematical model for the design of a filter to obtain the improved signal-to-noise (SNR) for the different audio signal ranges that can be used to strengthen the received signal and results high the power density. This paper presents Li-Fi filter that has been implemented for various audio frequency signals. The results reveal 2.4X, and 3.43X improvement in SNR for a frequency range of 60-250 Hz, and 2-3 kHz respectively. The presented filter is well suitable for Li-Fi-based audio transmission of human voice-related frequency for real-world applications including internet of things (IoT).
Keywords: *Audio frequency filter, Light Fidelity, Signal to noise ratio, Visible light communication, Internet of things*

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I. Introduction

Li-Fi (known as light fidelity) focuses on visual light communication (VLC), which employs light-emitting diodes (LEDs) to network wireless systems. It is a sub-domain of VLC technology that substitutes cable wire communication with light from LED or illumination lamps as a medium of communication. It enables electronic devices to connect to the internet at a stellar speed without any use of connecting cables. Li-Fi is similar to VLC in terms of medium of communication i.e. light, however the Li-Fi technology is fully networked, bidirectional, and extremely fast, whilst the latter only offers unidirectional, point-to-point light communication at a very high data speeds of communication. Also, it is advantageous over Wi-Fi as having almost no limitations in terms of data capacity since the visible light spectrum is around ten thousand times larger than the radio frequency (RF) spectrum. However, in confined spaces like factories, homes, offices, etc., we can achieve flexibility, security, wide coverage, and high throughput for both mobile and fixed devices with the help of 5G and Li-Fi [1]. The major disadvantage of Li-Fi is line of sight communication but, the laws of reflection and refraction allows it as a useful technology for applications like underwater and space communications.

Wireless channels are used to transmit the light signals to the receiver, which then translates the optical signals to recover the message. This technology is more secure than Wi-Fi as all of the information is encoded in light only which can be well contained in a physical space [2]. The standard *IEEE P802.15.13* is capable of transferring data up to *2.192 Gbps* in the range of *200 m* unrestricted line-of sight. New technologies which were not a part of the standard *IEEE P802.15.7* offer up to *10 Gbps* short-range speed using red-, green-, blue-, and yellow-colored LEDs [3]. With differing frequency responses, audio filters function as passive circuits or as amplifiers. These filters are an essential component of the fundamental building blocks of an audio system, much like microphones and speakers. They have the ability to either intensify or weaken a variety of audio input's frequencies. These filters, however, differ from a straightforward audio amplifier or input source because they act in a frequency dependent manner. No matter what frequency the incoming audio wave is at, it is amplified completely. A frequency-dependent amplifier, an audio filter operates between *0 Hz* and more than *20 kHz*. An optimized and specific amplification or attenuation of a range of frequencies in the audio signal paves the path to enhance the audio input tone [4].

In this work, the focus is on enhancing the quality of the signal and modulating it for a better result through simulation by creating more efficient sets of band pass filters to remove the noise from the signals and eventually improve the signal to noise ratio (SNR) so that it can be further used for enhancing the result of the previous as well as future research. Our previous work, observation, and simulation of power density and SNR at different heights and angles were done by us, and concluding with the most efficient value at 64.5 degrees and at 3.15m height of 8 dB/m [5]. Receiving the data signal without any distortion is one of the greatest challenges in digital signal processing. It is crucial to increase the signal's performance and lower the attenuation that random noise causes. Filters are mostly used in digital signal processing to reduce noise, which improves signal performance and aids in extracting the necessary data from the sent signal. Different filters are used to reduce the effect of noise [6]. Multiple filters can be operated in digital fashion over wide range of frequencies parallelly in the software simulation, the case unlikely in analog filters [7]. Some generalized matched filter designs earlier have demonstrated that for a weak input signal in noisy circumstances, SNR could be improved by inducting mutually independent noise components i.e. filter design using optimal noise [8].

II. Filtering And Data Enhancement

Digital signal is not a natural phenomenon; it is generated by sampling the analog signal into a digital signal. The final digital signal is produced by uniformly sampling the analog signal at a Nyquist rate ($f_{smax} = 2 \times f_{max}$), quantizing it, and encoding it. An extended high pass filter (HPF) is used for the audio filtering [9]. Various methods of audio filtering have been developed, including spectrum subtraction, Dolby noise reduction, the use of low-pass and high-pass filters, finite impulse response (FIR), and infinite impulse response (IIR) filtering [10]. A number of equations and difference formulae have been created to effectively perform time-varying filter applications. Numerous types of filters have been developed and suggested by numerous researchers to reduce noise in the ECG signal. The outcomes are simulated in Matlab framework [12] using an analog continuous time signal as the information signal. It is necessary to use uniform sampling to reduce signal loss [13].

In analog and digital communications, a signal-to-noise ratio, often known as SNR or S/N, is a measurement of the intensity of the desired signal in relation to background noise (unwanted signal). SNR can be determined using a specified formula that compares the two levels and returns a ratio. This ratio indicates whether the noise level is having an effect on the desired signal. The ratio is typically expressed as a decibel (*dB*) value. The ratio's value could be zero, positive, or negative. When the signal-to-noise ratio is higher than *0 dB*, the signal level is higher than the noise level. At greater SNR values, the quality of the signal improves [14].

III. Result And Discussions

As in real life situations the channel faces the random noise signal (thus we add noise file to the music file). The graph below depicts the original signal and noise signal. It is very easy to import an audio file in .wav format and plot its graph by mentioning the time to 1: end-1. As from the graph, it can be seen how the original analog signal is disturbed by the noise. When the audio signal is sampled as per the Nyquist rate, the added noise is causing inaccurate sampling. The encoded signal after sampling and quantization is disturbed with the addition of noise. The task is to lessen the noise component and retrieve back the information as present in the original audio signal. The graph of noisy signal (Audio + Noise signal) is shown in Fig. 1.

To reduce the noise from encoding along with information, the digital signal is passed through a digital band pass filter. This allows all frequencies other than the cutoff frequencies to be passed and other ones to be attenuated thus reducing the high frequency noise from the signal.

The transfer function of the used band pass filter is given by:

$$H(z)=\frac{A(z)}{B(z)}=\frac{b(1)+b(2)z-1+\cdots+b(n+1)z-n}{a(1)+a(2)z-1+\cdots+a(n+1)z-n}$$

In order to design a more efficient audio filter, we first need both channel plots for determining the noise frequency in general. Fig. 2 (a) shows the right-left channel plot.

The sampling frequency divided by two gives the highest frequency that can be represented in our signal. The Nyquist frequency is referred to as this. Our audio file's sample frequency was 48000 Hz, which indicates that the highest frequency it can represent is 24000 Hz. For each output element, FFT computation will be computed effectively using FFT function, which will need far less operations.[15]. In order to view the frequency spectrum of the signal, signal is simulated using FFT. The first parameter of FFT is the input signal, and the second one is the number of points to be evaluated. Only one channel has been taken into account while plotting the frequency spectrum in order to keep things simple as the other channel is identical. This is used as the FFT's initial input. In order to normalize the signal, findings are divided by N [16]. As seen in the frequency spectrum of Fig. 2(b), we applied the FFT shift to make the center map to 0 *Hz*, while the left ranges from 0 to -24000 *Hz* and the right extends from 0 to 24000 *Hz*. Negative frequencies might be thought of as those that go in the opposite direction. A negative frequency's frequency distribution should ideally match to that of a positive frequency. How much that frequency makes up the output is shown on a frequency spectrum graphic. The signal's magnitude, which is calculated by taking the abs function, defines its contribution [17]. The result is plotted in Fig. 3(b)

The FFT analysis of the given signal gives us the fundamental frequency and harmonics (say up to n^{th} harmonic) that are crucial for intelligibility and a quality reconstruction of the audio. Based on the plot, the pass band cut off frequencies (f_L and f_H) are selected slightly wider than the nth harmonic frequencies of the audio signal i.e. let's say the frequencies of the signal lies in range [f_a, f_b] then select $f_H < f_a$ and $f_L > f_b$, rejecting all noise frequencies beyond this boundary. Thus, the filter involves an automated FFT calculator and a band pass filter with variable cutoff frequencies filtering all the noise and distortion elements out of the signal.

IV. Implementation Of Filter Design For Audio Frequency

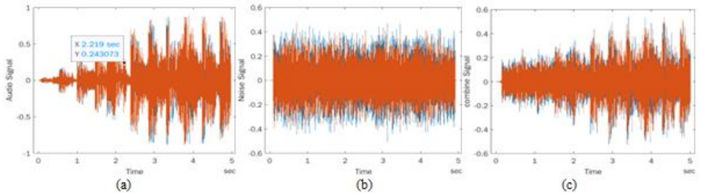

Figure 1. (a) Sample audio signal (b) Sample noise signal (c) Sample noise audio signal

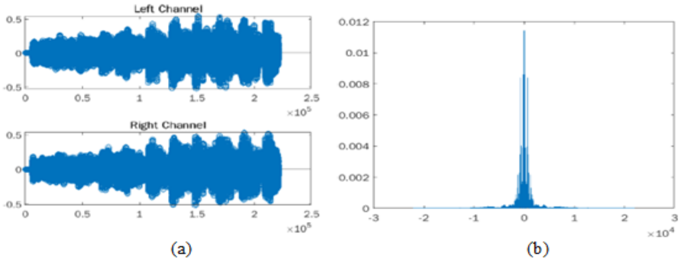

Figure 2. (a) Sample audio signal (b) sample noise signal (c) Sample noise audio signal

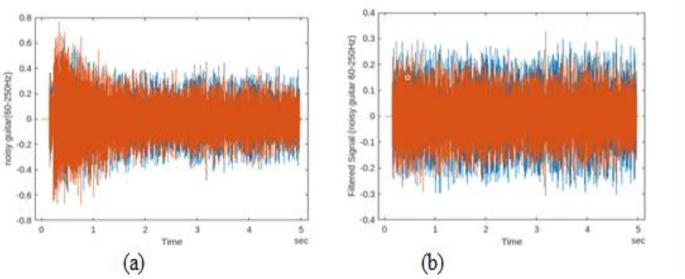

Figure 3. (a) Noisy and (b) filtered signal containing music as well as noise for 60-250 Hz

Table 1. SNR analysis

Frequency range	SNR before filtering	SNR after filtering	Improvement in SNR
60-250 Hz	-6.1838 dB	- 2.5763 dB	3.6076 dB
2-3 KHz	3.1894 dB	10.9655 dB	7.7761 dB
6-8 KHz	5.7981 dB	0.0598 dB	5.7383 dB

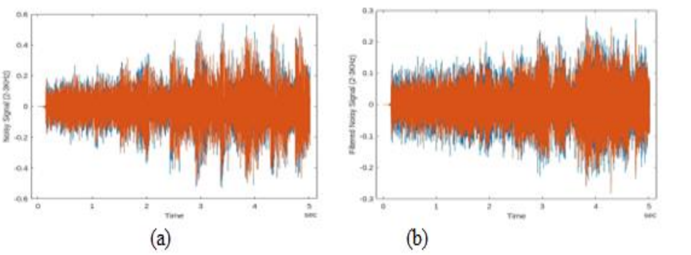

Figure 4. (a) Noisy and (b) filtered signal containing music as well as noise for 2-3 kHz

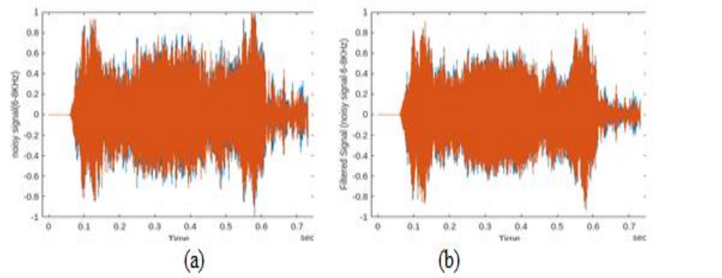

Figure 5. (a) Noisy and (b) filtered signal containing music as well as noise for 6-8 kHz

As mentioned above, a band-pass filter is implemented on three different sets of guitar frequency and compared the plot of filtered signals for each of the files. As the result of the SNR calculated for each range of frequencies, we have arranged the calculated data in a schematic way of SNR before and after the implementation of the filter and gathered it in a tabular form below. The results reveal 2.4X, and 3.43X improvement in SNR for frequency ranges of 60-250 Hz, and 2-3 kHz respectively. Taking human speech in context, the frequency range between 1 kHz and 4kHz of human speech is of the most importance for intelligibility [18] and standard telephony has been limited to the frequency range below 4 kHz [19] thus, proving the above designed filter to be very efficient in this frequency range with improvement in SNR of up to 7.77dB. Also, for the most widely used musical instruments, the fundamental frequencies lie usually in the range of 30 Hz - 3KHz [20]. The above filter is to be employed in designing of the trans- receiver system of Li-Fi to enhance the signal transfer and remove distortions due to noise if any.

V. Conclusion And Future Scope

This article shows the work and importance of filters in the Li-Fi technology. We extended our previous works on various factors affecting the data transmission to design a filter. In this work, a .wav file with noise is used and filtered using software techniques to remove undesired noise and convert it into a noise-free signal and appropriate graphs are plotted to know the contrast in frequency of the noisy signal and filtered signal after various simulation with different frequencies and voice. The results/ calculation shows the capacity of SNR before and after the filter and most importantly the improvements in them. The most capable frequency range is 2-3 kHz which is the general human voice frequency. This work aimed to maximize the signal/noise ratio and reduce distortions. Digital filters despite having benefits of flexibility and automatic adjustment using software, however pose certain disadvantages such as signal level affected upon the conversion from the digital to the analog domain and also requirement of a greater number of bits for filter response calculations and to prevent overflows [21]. For future work, we will aim to develop better loudness/power ratio (probably through controlling directional sound better), also cheaper and lighter amplifiers. Also, a higher frequency range will be covered for more efficient data-intensive applications.

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