Designing communication systems for BASA

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EGB242 : Signal Analysis

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Introduction

BASA is in the process of implementing a communication system to establish seamless correspondence with the MARS-242 orbiter. To accomplish this, it is imperative to effectively denoise the signal received from a microphone stationed at the mission control center. This denoised signal must be effectively transmitted to the MARS-242 orbiter, where it will undergo demodulation to allow the crew to listen to transmissions. This report primarily focuses on three key objectives: eliminating periodic noise, managing signal transmission and reception, and performing analog-to-digital conversion.

Elimination periodic noise

After testing the microphone equipment at mission control, it has been observed that a periodic signal is interfering with the microphone audio.

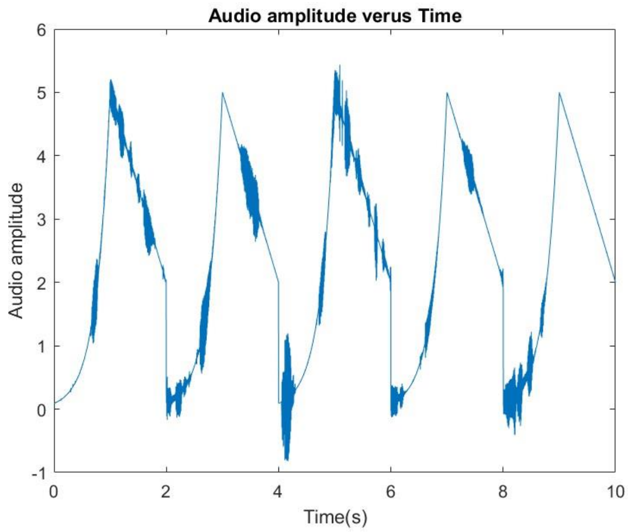

Figure 1: Audio Amplitude Verus Time, Noisy Audio signal.

In Figure 1 depicted above, a periodic signal with a 2-second cycle is causing interference with the microphone audio. This interference has led to a substantial amount of noise in the microphone's audio output being inaudible, rendering the signal incomprehensible. This can be seen within figure 1, given that audio can only be heard around the peaks visible peaks when time is odd(1,3,5,7 seconds). To address and mitigate this periodic noise, a model of the noise signal generated by the equipment was determined. The analysis revealed that the noise signal could be characterized by:

$$n(t) = \begin{cases} 5e^{4(t-1)} & 0 \le t < 1 \\ -3t + 8 & 1 \le t < 2 \end{cases}$$

Figure 2: Model of Noise Signal

This signal can be easily decomposed into a complex Fourier series with the coefficients characterized by the expressions:

$$C_0 = \frac{19 - 5e^{-4}}{8}$$
$$C_0 \approx 2.3635$$

And,

$$C_n = \frac{1}{2}\left(\left(\frac{5e^{-p} - 5e^{-4}}{4-p}\right) + \frac{(5pe^{-p} - 2p + 3 - 3e^{-p})}{p^2}\right)$$

Eq 1

Whereby $p = j\pi n$

Using these expressions, the first 5 harmonics of this noise signal can be calculated to be:

Table 1: First 5 positive harmonics (C_n) given by the expression given in equation 1.

	C_n					
n	0	1	2	3	4	5
Real(ℝ)	2.3635	-0.6976	0.1769	-0.1309	0.0564	-0.0509
Imaginary(ℂ)	0	0.8049	0.0392	0.1425	0.0580	0.0706

Table 2: First 5 negative harmonics (C_n) given by the expression given in equation 1.

	C_n					
n	0	-1	-2	-3	-4	-5
Real(ℝ)	2.3635	-0.6976	0.1769	-0.1309	0.0564	-0.0509
Imaginary(ℂ)	0	-0.8049	-0.0392	-0.1425	-0.0580	-0.0706

These values calculated, shown in table 1 and 2, can be subsequently used to form an approximation of the noise signal for the full-time vector using the equation:

$$s(t) = \sum_{n=0}^{\infty} C_n e^{jn\pi t}$$

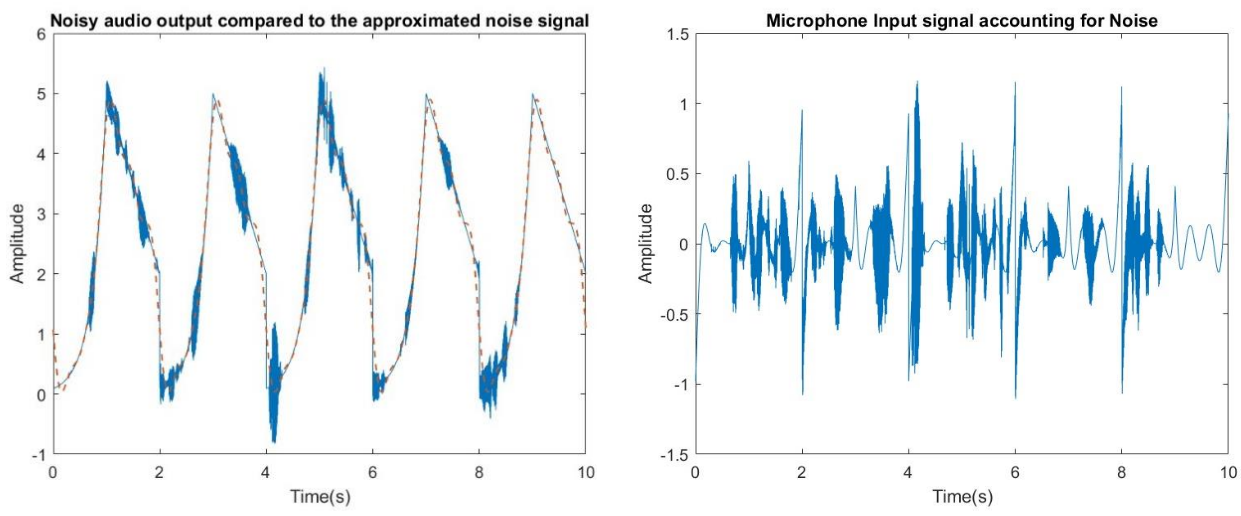

Figure 3: LEFT: The noisy audio output compared to 5^{th} order approximation of signal noise. RIGHT: The Microphone signal accounting for noise using 5^{th} order approximation of signal noise.

In Figure 3(LEFT), the 5^{th} order Fourier series approximation closely matches the audio profile. By reversing the additive noise process, a clean microphone input signal can be produced. Figure 3 (RIGHT) demonstrates the successful reduction in noise produced a relatively clean audio profile using a 5^{th} order approximation (utilizing 5 harmonics). Audio testing confirms the clarity of human speech. While equipment-induced noise is effectively removed, the static from the microphone itself becomes the prominent cause of audio interference. As a result, the 5th order approximation sufficiently reduces the noise caused by equipment and no noticeable improvement of audio quality can be identified using up to a 200th order approximation. When reduced to a first order approximation, audio is still clear and concise, however with a slight increase in audio pops (peaking audio). Thus a 5th order approximation is sufficient for the current BASA communications systems. To further enhance audio quality, the microphone-generated noise must be reduced. With noise effectively mitigated to our current capabilities, the signal is now ready for transmission from the ground station to the MARS-242 orbiter.

Managing signal transmission and reception

The signal produced from the microphone must be transmitted to the MARS-242 orbiter. To achieve this the signal must be modulated and centered around a carrier frequency to be transmitted. The magnitude spectrum of the clean audio signal was evaluated to determine the spectrum range of the input signal.

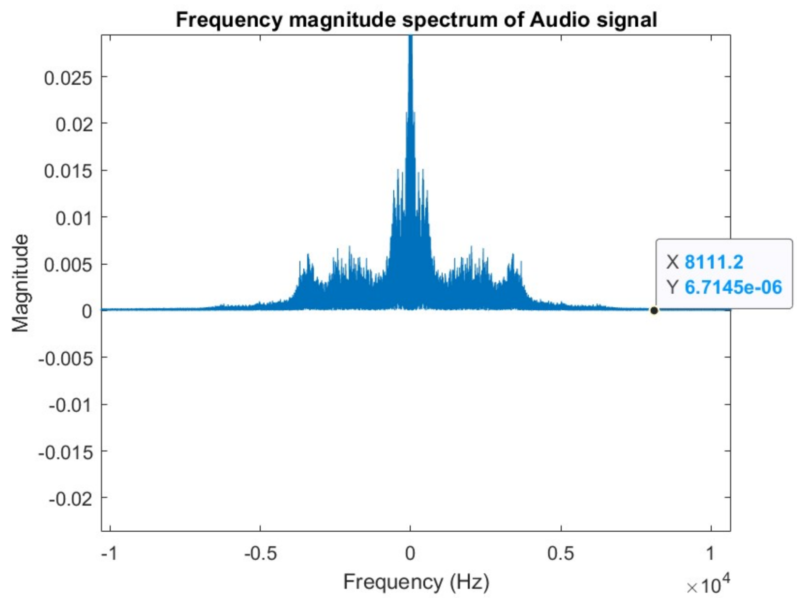

Figure 4: Frequency magnitude spectrum of the audio signal produced from the microphone

The frequency spectrum range comfortably sits within the frequency bounds of (-8100,8100) because the signal is centered on 0. This results in a total frequency range required for the transmission of the signal to be 16.2kHz.

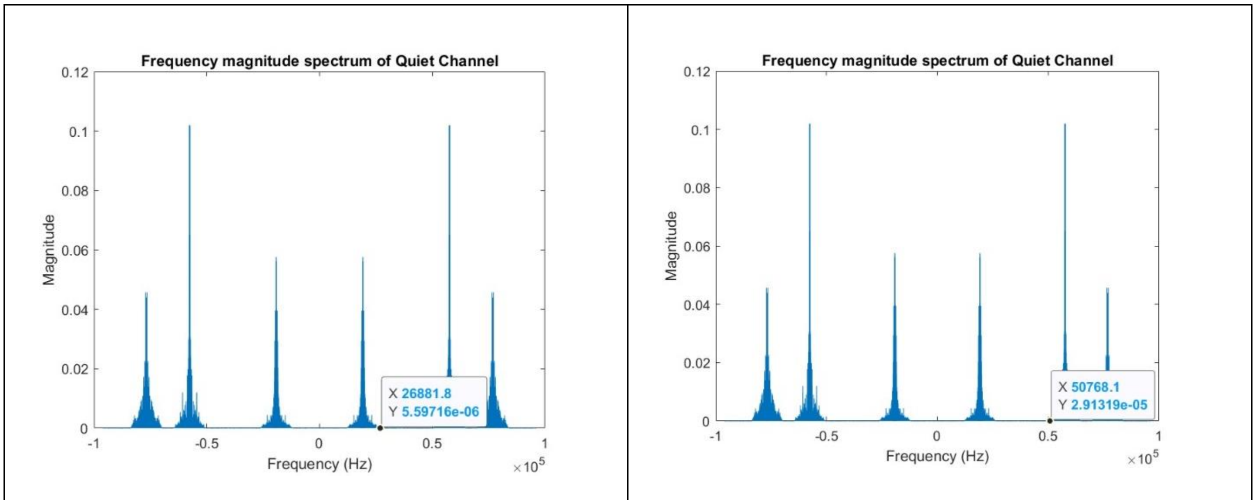

Figure 5: The upper and lower bounds for a potential frequency range to be used for signal transfer

To transmit the data, a signal channel is to be used. This channel has multiple other active frequencies thus an unused frequency range must be found. As seen in figure 5, a frequency range of 27kHz to 50kHz is unused. A similar frequency range is found on the positive range at -50kHz to -27kHz. The center is located at $\pm$ 38.5kHz around the 0 frequency. This allows for the $\pm$8.1kHz range around 38.5kHz (30.4kHz to 46.6kHz and -46.6kHz to -30.4kHz) which is unused and thus will be used for signal transfer. To send the signal at these frequency ranges, the signal must be modulated such that the signal center is located at 38.5kHz. This is achieved by the formula:

$$x(t)\cos(2\pi f_c t)$$

Which in our case is implemented as:

$$m(t) = s(t) * \cos(2\pi * 38500t)$$

Whereby, $s(t)$ is input signal and $m(t)$ is the modulated signal. This is simply implemented in MATLAB as;

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mod = audioClean.*(cos(2*pi*38500*t));
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Figure 6: Modulation calculation implementation in MATLAB

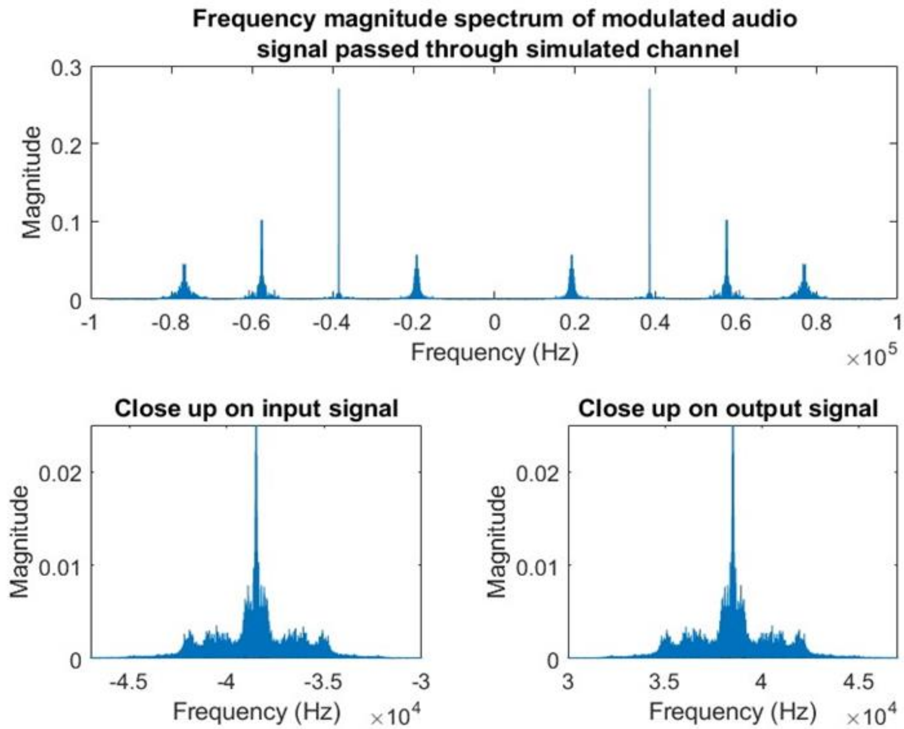


Figure 7: Modulated audio signal from microphone centered around ± 38.5kHz

As figure 7 depicts, the signal has been successfully modulated around ± 38.5kHz. It can be seen that the signal is still intact, and both the input and output signals do not clip any of the other actively used frequencies in use. The signal transmission has been successfully achieved; however the received signal must be demodulated in order to be converted back into an audio signal.

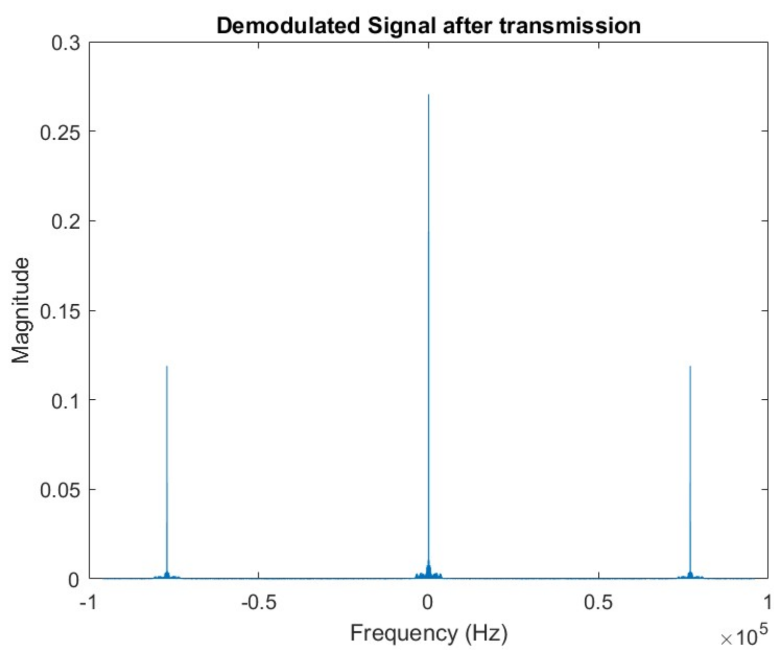


Figure 8: Demodulated Signal after transmission through the channel

To isolate the required auto signal, a lowpass filter was employed. This allowed for the isolation of the required audio signal by attenuating higher frequencies. This was achieved by using lowpass MATLAB function and resulted in an affective reconstruction of the transmitted signal. The other three signals occupying the channel frequency domain were also investigated. It was found that the audio signal centered on 19.2kHz is “Kiss From A Rose” by SEAL. The audio signal centered on 57.6kHz is a guitar riff and the audio signal centered on 76.8kHz is a guitar solo.

Analog-to-digital Conversion

The signal is able to be heard by the crew within the MARS-242 orbiter, however the orbiters communication systems require the digital audio signal to be digitized. Due to Nyquist's theorem, the sampling rate must be greater than twice the highest frequency component. As a result, the minimum sampling rate is 16.2kHz. The closest commonly used sampling frequency is thus 22.05kHz. This is to ensure the signal remains intact and without distortion. Using resample, the audio signal can be remapped to 22.05kHz thus allowing its use by common sampling rate computer systems. The resampled audio when played at the frequency of 22.05kHz is unchanged.

Conclusion

To summarize, this report has been a rigorous computation of communication systems between BASA mission control and the MARS-242 orbiter. The report had three principal objectives: the elimination of periodic noise, the transmission and reception of signals, and analog-to-digital conversion.

In addressing periodic noise, Fourier series analysis was effectively utilized to mitigate interference. The 5th order approximation of noise was able to significantly remove equipment noise interference. This resulted in microphone-generated static as the primary audio interference source.

Regarding signal transmission and reception, the employed modulation ensured no interference with active signals within the channel. Modulating the signal around ±38.5kHz enabled effective signal transmission, with subsequent demodulation for signal extraction using a low pass filter. The previously employed frequency ranges were investigated discovering; “Kiss From A Rose” by SEAL at 19.2kHz, a guitar riff at 57.6kHz and guitar solo at 76.8kHz.

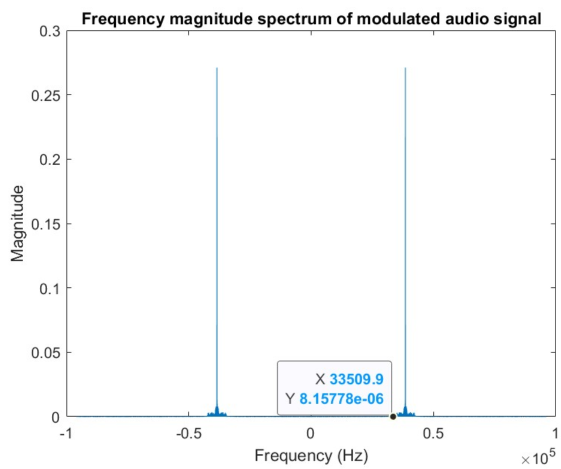
In analog-to-digital conversion, despite being unable to complete this section, it was determined that adherence to Nyquist's theorem was pivotal. Resampling the audio signal to 22.05kHz ensures compatibility with common computer systems. This allows for the signal to be quantized by common computer systems.

Reflection

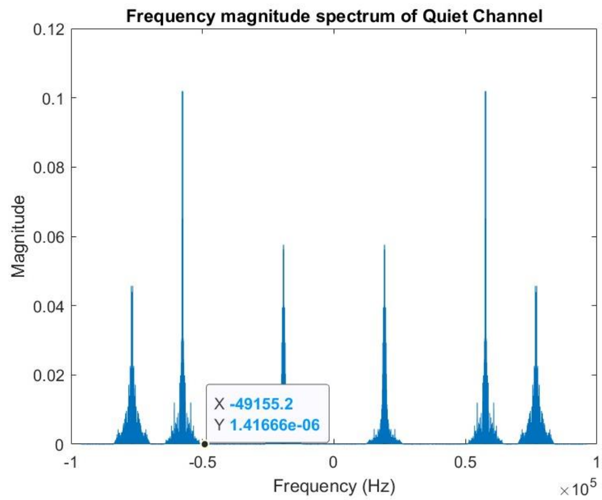
I demonstrated a solid understanding of communication systems engineering. I effectively eliminated periodic noise from a microphone's audio signal using Fourier series analysis completed by hand, ultimately determining that a 5th order approximation was almost more than sufficient. I then was able to modulate and transmit the cleaned audio signal, through a carrier frequency while avoiding interference with other active frequencies. The subsequent demodulation process and isolation of the signal allowed for the signal from the microphone to be reconstructed at the MARS-242 orbiter. Finally, I addressed analog-to-digital conversion, ensuring the signal could be saved, if quantized. These skills demonstrate my ability in key communication engineering concepts, required for designing effective communication systems, such as BASA's communication system with the MARS-242 orbiter.

This assignment, regrettably, ended up being hastily executed and only partially completed owing to my own time management skills and noticeable gaps in my knowledge base. I significantly underestimated the time commitment required and subsequently found myself grappling with the challenge of completing it within my set timeframe. This was primarily a result of a combination of procrastination and a lack of motivation. However, despite the rushed nature of my efforts, I found that this assignment offered significant value in terms of filling in the gaps within my understanding of real-world communication systems. This experience has served as a catalyst, assisting my motivation to enhance my proficiency and attain a deeper level of knowledge within this field.

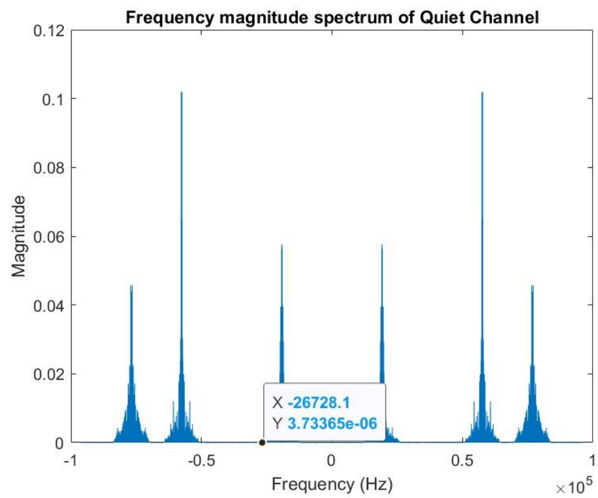
Appendix



Appendix 1: Lower bound after modulation



Appendix 2: Lower Bound of available frequency range in channel



Appendix 3: Upper bound of frequency range available in channel