WACS

Wireless Audio Communication System

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Group 10

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1 Summary

During this project a digital wireless communication system consisting of a transmitter and a receiver has been implemented. The system performs encoding/decoding, modulation/demodulation and band limitation with specifications determined both based on given restrictions and through testing. The system was tested by repeatedly transmitting and receiving messages of varying lengths. For shorter messages, the system has a low error rate and returns the full message. However, when sending a long message it is frequent that half of that message is corrupted. During testing, it was also shown that the system was sensitive to background noise and performed significantly worse in noisy environments.

2 Introduction

The purpose of the project is to implement a digital wireless communication system that is able to transmit data between computers using audio signals. The system should be able to synthesize simple wireless communication signals using sampling, filtering, modulation and demodulation.

The primary objective is to design and analyze the components of the communication system, the transmitter and receiver, with regard to certain requirements. The system needs to adhere to the assigned frequency band, between 3475Hz and 3525Hz, and the power restriction of 27dBm [1], in order not to infringe on communications carried out on other frequencies.

The system's performance is measured by the ability to transmit and receive data accurately, not just under perfect conditions but also when subjected to background noise and with other signals being transmitted at the same time. The messages received by the receiver should in the ideal case remain the same as the messages that were transmitted.

3 Theory

The system consists of a transmitter and a receiver. The transmitter encodes a message, modulates it, and ensures that the transmission complies with the requirements to avoid interference with other communication systems by applying band limitation. The receiver listens to the specific frequency band, demodulates and finally decodes the received message.

The system needs two types of filters to achieve this functionality; a band-limiting filter for both the transmitter and receiver, and a low-pass filter for the demodulator to correctly process the signal.

3.1 Modulator and demodulator analysis and design

One of the main features of the system is the modulation and demodulation processes. The baseband signal, $x_b(t)$, cannot be directly transmitted wirelessly, but first has to be shifted to be centered around the carrier frequency, $f_{carrier}$. In this system, the modulation process used is Binary Phase Shift Keying (BPSK), which encodes the symbols using different phase shifts of the carrier signal. The phase shift added is $-\pi$ when $x_b(t) = -1$ and 0 when $x_b(t) = 1$. -1 is the representation of a binary zero, and 1 of a binary 1.

The spectrum of the modulated signal $x_m(t)$ can be seen in Equation 1. The equation describes sums of sinc functions shifted with ω_c and $-\omega_c$. The equations confirm how effectively the system uses the available bandwidth and aids when evaluating parameters such as the symbol duration T_b.

$$
\begin{aligned}
X_m(\omega) &= \frac{1}{2\pi}\mathcal{F}\{x_b(t)\} * \mathcal{F}\{A_c sin(\omega_c t)\} = \\
&= \frac{1}{2\pi}(\int_{-\infty}^{\infty}(T_b\sum_{n=0}^{N-1} b_n\,\mathrm{sinc}(\frac{\tau T_b}{2\pi}e^{-j\omega\frac{(2n+1)T_b}{2}})) \cdot A_c j\pi[\delta(\omega-\tau+\omega_c)-\delta(\omega-\tau-\omega_c)]d\tau) \\
&= \frac{T_b A_c b_n j}{2}(\sum_{n=0}^{N-1}\mathrm{sinc}(\frac{(\omega+\omega_c)T_b}{2\pi})e^{-j(\omega+\omega_c)\frac{(2n+1)T_b}{2}} - \sum_{n=0}^{N-1}\mathrm{sinc}(\frac{(\omega-\omega_c)T_b}{2\pi})e^{-j(\omega-\omega_c)\frac{(2n+1)T_b}{2}}) \\
&= \frac{A_c j}{2}(X_b(\omega+\omega_c) - X_b(\omega-\omega_c))
\end{aligned} \tag{1}
$$

The assigned frequency band is $[3475, 3525]$Hz. The carrier frequency $f_{carrier}$ was therefore chosen to be 3500Hz, precisely in the middle of the band, ensuring that the system can use the entire available frequency range. When utilizing the full frequency band the system can support a shorter symbol duration T_b and therefore a higher bitrate, meaning the system can transfer data faster. If the carrier frequency was chosen closer to one of the ends of the assigned frequency band, the symmetric length to each side would be smaller, thereby not allowing effective use of the full frequency band. This would place further limitations on the design of the system to avoid interfering with adjacent frequency bands.

When choosing the bandwidth time T_b it is taken into account whether the system should process one or two sidelobes in addition to the main lobe of the signal. Using the fact that the total width of the assigned frequency band is 50Hz (bandwidth), and the width of a main lobe (given by the formula sheet) is $2 \cdot \frac{1}{T_b} = \frac{2}{T_b}$, and the width of the sidelobes is $\frac{1}{Tb}$, a suitable T_b can be calculated. This is calculated according to Equation 2:

$$
T_b = \frac{2 \cdot (N+1)}{\text{bandwidth}}, \tag{2}
$$

where N represents the number of sidelobes on one side of the main lobe. This gives $T_b = 0.08$ for 1 sidelobe and $T_b = 0.12$ for 2 sidelobes.

Consequently, T_b should fall within the interval $[0.08; 0.12]$ where the exact choice will affect the bitrate. The bitrate R of the system is determined by Equation 3, where for $T_{b,1} = 0.08$, $R_1 = 12.5$bits/s and for $T_{b,2} = 0.12$, $R_2 \approx 8.33$bits/s.

$$
R = \frac{1}{T_b} \tag{3}
$$

Having more sidelobes within the assigned frequency band means choosing a larger T_b, which in turn will lead to a lower bitrate. The advantage of choosing a larger T_b (and accepting a lower bitrate), is the reduced risk of smearing and thereby interference with other frequency bands, since the main lobe is more tightly concentrated around the carrier frequency, as seen in Figure 1 when compared with fewer sidelobes. A smaller T_b and a higher bitrate will instead allow the system to process data faster, but have an increased risk of inference since the larger main- and sidelobes extend closer to the edges of the frequency band. For this system, robustness and stability were prioritized over speed, which led to the decision to include two sidelobes and, hence, a lower bitrate.

The maximum amplitude A_c of the carrier signal $f_{carrier}$ must be chosen so as not to violate the energy constraint of 27dBm. Using Equation 4, the amplitude can be determined as $A_c \approx 1$.

$$
P = 10^{\frac{P_{dBm}}{10}} = \frac{A^2}{2} \Longrightarrow A = \sqrt{10^{\frac{P_{dBm}}{10}} \cdot 2} \tag{4}
$$

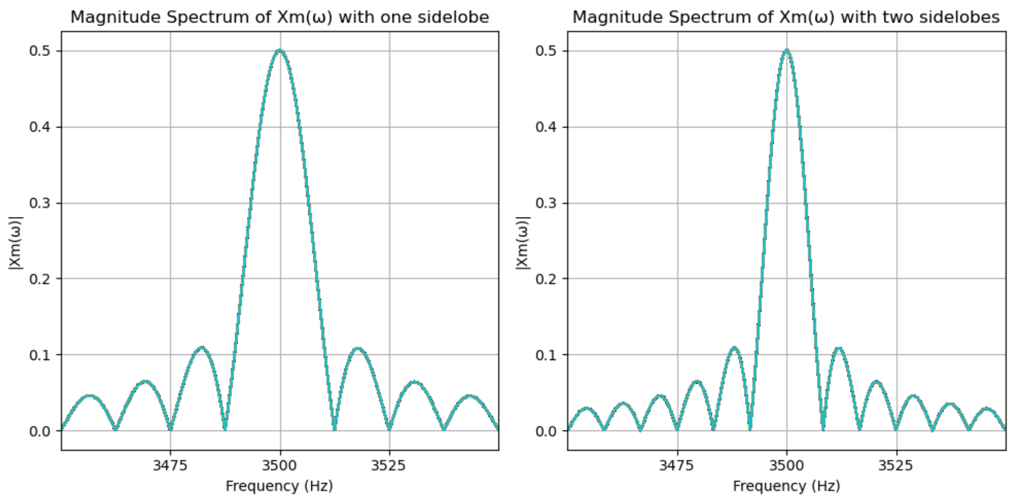

Figure 1: One sidelobe and two sidelobes inside the assigned frequency band

The transmitter modulates the signal to make it possible to transmit it, which means that the receiver needs a demodulator to reverse the modulation process in an attempt to recover the original baseband signal.

The first step of the demodulation is the down-shift in frequency, meaning that the frequency components centered around the carrier frequency are moved to be centered around $\omega = 0$. This is done using in-phase quadrature (IQ) demodulation, which involves splitting the signal into two components by multiplying it with a sine and cosine signal of the carrier frequency. This means that the demodulated baseband signal $y_b(t)$ can be described as seen in Equation 5 with the frequency components from Equation 6a and 6b. The spectrums $Y_{I,d}(\omega)$ and $Y_{Q,d}(\omega)$ are derived in Equation 7.

$$y_b(t) = y_{I,b}(t) + jy_{Q,b}(t). \tag{5}$$

$$\begin{aligned} y_{I,d}(t) = y_m(t)cos(\omega_c t) &= A_r y_b(t-t_0)sin(\omega_c t + \varphi_r)cos(\omega_c t) \\ &= \frac{A_r}{2} y_b(t-t_0)(sin(2\omega_c t + \varphi_r) + sin(\varphi_r)) \qquad (6a) \\ y_{Q,d}(t) = -y_m(t)sin(\omega_c t) &= -A_r y_b(t-t_0)sin(\omega_c t + \varphi_r)sin(\omega_c t) \\ &= -\frac{A_r}{2} y_b(t-t_0)(cos(\varphi_r) - cos(2\omega_c t + \varphi_r)) \qquad (6b) \end{aligned}$$

$$\begin{aligned} Y_{I,d}(\omega) &= \frac{A_r}{2}(\frac{1}{2\pi}(Y_b(\omega)e^{-j\omega t_0} * \frac{1}{2j}(e^{j\varphi_r}2\pi\delta(\omega - 2\omega_c) - e^{-j\varphi_r}2\pi\delta(\omega + 2\omega_c)) + Y_b(\omega)e^{-j\omega t_0}sin(\varphi_r)) \\ &= \frac{A_r}{4j}(Y_b(\omega - 2\omega_c)e^{-j(t_0(\omega - 2\omega_c) - \varphi_r)} - Y_b(\omega + 2\omega_c)e^{-j(t_0(\omega + 2\omega_c) + \varphi_r)} + Y_b(\omega)e^{-j\omega t_0}sin(\varphi_r)) \end{aligned} \tag{7a}$$

$$\begin{aligned} Y_{Q,d}(\omega) &= -\frac{A_r}{2}(\frac{1}{2\pi}(Y_b(\omega)e^{-j\omega t_0} * \frac{1}{2}(e^{j\varphi_r}2\pi\delta(\omega - 2\omega_c) + e^{-j\varphi_r}2\pi\delta(\omega + 2\omega_c)) + Y_b(\omega)e^{-j\omega t_0}cos(\varphi_r)) \\ &= -\frac{A_r}{4}(Y_b(\omega - 2\omega_c)e^{-j(t_0(\omega - 2\omega_c) - \varphi_r)} + Y_b(\omega + 2\omega_c)e^{-j(t_0(\omega + 2\omega_c) + \varphi_r)} + Y_b(\omega)e^{-j\omega t_0}cos(\varphi_r)) \end{aligned} \tag{7b}$$

However, this process will also create a new high-frequency component at a frequency that is double the carrier frequency and is seen through the shift of $2\omega_c$ in Equation 7. To get rid of this, the signal will also have to be low-pass filtered. The design of the low-pass filter is discussed in Section 3.2.2.

3.2 Filter specification

Two types of filters are needed for the system. Firstly, a band-limiting filter is needed for the transmitter to make sure that the system adheres to the requirement of staying within the allowed frequency band, and for the receiver to filter out any noise that lies outside of the relevant frequency band. Secondly, a low-pass filter is needed for the demodulator to remove the high-frequency component that is introduced by the IQ demodulation process.

3.2.1 Band limitation filter

A suitable band-pass filter for the transmitter and receiver would have passband edge frequencies 3475Hz ($f_{pass,1}$) and 3525Hz ($f_{pass,2}$), the edges of the assigned frequency band, and stopband edge frequencies 3450Hz ($f_{stop,1}$) and 3550Hz ($f_{stop,2}$). The choice of frequencies supports a transition band of 25Hz in either direction of the assigned frequency band, avoiding possible interference from adjacent frequency bands. With a passband ripple of 1dB (A_{pass}) and a stopband attenuation of 60dB (A_{stop}), the system can properly attenuate signals outside and inside the frequency band. The attenuation values were chosen through testing and are discussed and motivated further in section 4.2.2.

The most optimal filter is a filter having unbound attenuation and no ripples in the passband. This would result in having minimum disturbance of the signal while also filtering out as much of the background noise as possible. This is a characteristic of a Butterworth filter. However, due to the small transition band of 25Hz, this filter type would make the system unstable and unusable. The second most optimal filter is a Chebyshev type 2 filter, which has no ripples in the passband with limited attenuation. However, it also makes the system unstable due to the small transition band in combination with the small passband of 50Hz. Because of this, an elliptic filter was chosen as it is the only filter type with a high enough roll-off. To give further proof, all three filter types were tested, and the elliptic filter was the only stable one. A sketch of the band limiting filter can be seen at the top of Figure 2.

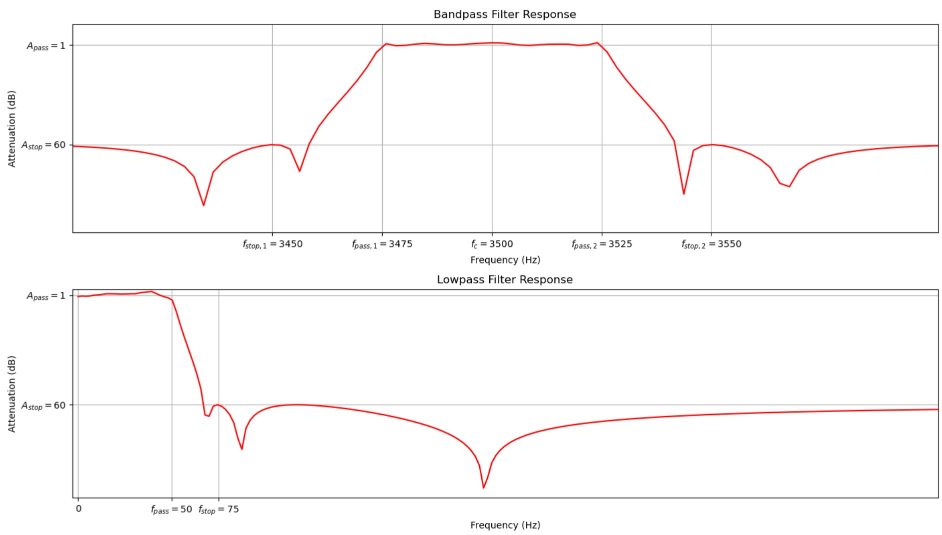

Figure 2: Sketch of the implemented band-limiting filter (top) of the system with passband edge frequencies, stopband edge frequencies, as well as a passband ripple and stopband attenuation. Sketch of the implemented low-pass filter of the system (bottom) with passband edge frequency, stopband edge frequency as well as passband ripple and stopband attenuation.

3.2.2 Demodulation low-pass filter

The bandwidth of the demodulated baseband signal, 50Hz, means that the low-pass filter should have the passband edge frequency 50Hz, to not filter out any of the signal. The stopband edge frequency chosen for the filter is 75Hz, which gives a short transition band of 25Hz and filters out background noise in addition to removing the introduced higher frequency component. The passband ripple was set to 1dB to minimize the distortion in the passband, and the stopband attenuation was set to 60dB, to effectively suppress any noise from unwanted frequencies, and is also further discussed in section 4.2.2. The implemented low-pass filter is sketched at the bottom of Figure 2.

3.3 Sampling frequency

The continuous signal needs to be sampled to a discrete-time signal in order to be processed by a computer. The sampling frequency should not violate the Nyquist-Shannon sampling theorem and also be chosen so that the system can obtain a pure sine signal. To achieve this, the sampling frequency should fulfill the two conditions in Equations 8a and 8b, where K_c is the period of the discrete-time carrier signal and should be an integer to ensure a pure sine signal.

$$\omega_s \gg 4\omega_c \tag{8a}$$

$$K_c T_s = T_c \tag{8b}$$

The Nyquist frequency f_N can be set equal to the carrier frequency, $f_N = f_{carrier} = 3500\text{Hz}$, which gives $f_s \gg 4f_N = 14\text{kHz}$, according to Equation 8a. That is, the smallest sampling frequency suitable for the system is 14kHz. To have a larger range and minimize the risk of aliasing, the sampling frequency chosen for the system is $10 \cdot f_N = 35\text{kHz}$.

Hence, the sampling period $T_s = \frac{1}{f_s} = \frac{1}{35k} \approx 2,86 \cdot 10^{-5}s$, which for Equation 8b gives $K_c = 10$.

4 Results and discussion

4.1 Test results

Tests were carried out by using three different test sentences and transmitting each sentence 10 times, with different passband- and stopband attenuations. The test sentences were

- **"a"** (1 character, 8 bits)
- **"daffodilly"** (10 characters, 80 bits)
- **"Lorem ipsum dolor sit amet, consectetur adipiscing elit. Nulla sit amet aliquet felis. Nulla non tur"** (100 characters, 800 bits)

and the results of the tests can be seen in Table 1.

Table 1: Testing Table

Characters (bits)	**1** (8)		**10** (80)		**100** (800)	
Bit averages	Incorrect	Not received	Incorrect	Not received	Incorrect	Not received
$A_p = 3\text{dB}$ $A_s = 60\text{dB}$	77.5%	45%	100%	56.5%	53.5%	11.8%
$A_p = 1\text{dB}$ $A_s = 60\text{dB}$	0%	12.5%	18%	7.1%	45.5%	2%
$A_p = 0.1\text{dB}$ $A_s = 60\text{dB}$	Unstable					
$A_p = 3\text{dB}$ $A_s = 40\text{dB}$	22.5%	60%	25.5%	4%	46.9%	2.3%
$A_p = 1\text{dB}$ $A_s = 40\text{dB}$	0%	12,5%	4.5%	2.3%	57.3%	0.1%
$A_p = 0.1\text{dB}$ $A_s = 40\text{dB}$	0%	11.3%	0%	2.5%	56.4%	0.1%

Based on the tests, the chosen combination of attenuation values is 1dB for the passband and 60dB for the stopband. This is mainly because that combination gave the best results for longer messages. Important to note is that these tests were conducted in a relatively quiet environment, a closed room with minimal background noise. When attempting to transmit the same messages in environments with significant background noise, including other high-pitch signals, the error rate increased and the number of correctly received bits was lowered significantly.

4.2 Errors

Errors such as bit misinterpretation, missed bits or background noise being misread as bits are some of the possible errors that can occur when using the system. Misinterpreted bits results in corrupted symbols, due to the binary value now representing another symbol. Missing a bit shifts all following bits, thus each bit will be interpreted as part of the previous symbol, corrupting the entire message following the missed bit. Similarly, an extra bit being picked will lead to the rest of the message being corrupted, after the added bit.

4.2.1 Error rate - Missing bits

An error that consistently occurred, regardless of message length or attenuation values, was that one or two bits would be missing at the end of the message, causing the last character to be incomplete or misinterpreted. This was traced to a frequency shift in the frequency domain, corresponding to a time shift in the time domain, caused by the frequency-domain filtering in the transmitter. Adding zero-padding to the end of the transmitted signal before filtering resolved the issue, ensuring that all bits were correctly delivered, and no parts of the message were mistakenly filtered out. Additional tests were carried out after the zero-padding was added, which confirmed that the issue was resolved, showing the expected behavior of all bits being received. It was concluded that future tests would only affect bit count without changing the overall results, and thereby conclusions made from the results, in Table 1. Therefore, the test data was not redone with added zero-padding.

4.2.2 Error rate - Attenuation

Errors can also be a result of poorly chosen filter attenuation. High passband attenuation A_{pass} may cause the signal to be missed due to low energy. Similarly, low stopband attenuation A_{stop} can allow background noise to interfere, causing all three types of corruptions mentioned in 4.2.

As seen in Table 1, error rates depend on the attenuation in both passband and stopband. Higher attenuation in the passband lowers the energy of the signal and can increase the chance of missing signals. However, higher attenuation in the stopband reduces background noise and makes it less likely to be interpreted as part of the signal, reducing interference.

Lower stopband attenuation works better in conjunction with other audio communications systems. With other systems transmitting with similar or the same frequency components, interference might occur. To ensure that more of the signal is picked up, a lower attenuation is preferred. This is due to less of the corrupted signal being filtered out. However, when testing with other systems simultaneously, there was no noticeable difference between using 40dB and 60dB in stopband attenuation.

4.2.3 Error rate - Message length

As seen in Table 1, the error rate varies with the length of the message. When sending a message, there is a chance that an error occurs at any of the bits sent. Thus, if the message is longer and contains more bits, the risk of error becomes larger. This includes all three types of the mentioned errors in 4.2. Another cause for error could be desynchronization between clocks of the transmitter and receiver.

5 Conclusion

The system for wireless communication that was created is functional and somewhat resistant to interference and noise. The design of the system was focused on robustness and creating a system as reliable as possible. This meant making trade-offs in other aspects, for example, having a lower bit rate. Despite the focus on making the system stable, it is still quite sensitive to noise, especially when interfered with by frequencies close to the frequency band used by the system. This is mainly due to the relatively low bandwidth, and short margins to adjacent frequency bands, which put a lot of restrictions on the design of the system.

References

[1] *Project in signals and transforms wireless audio communication system*, https://uppsala.instructure.com/courses/97637/files/7578141?wrap=1, Accessed: 2024-12-20.