Analog Communication System Design: Project Report

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Introduction

Analog communication systems serve as fundamental components in the transmission of information, enabling reliable and efficient communication. This project focuses on designing and implementing an analog communication system using Simulink, to transmit two recorded voice messages simultaneously through both amplitude modulation (AM) and frequency modulation (FM) techniques. The system operates around selected carrier frequencies and is designed to mitigate the effects of additive white Gaussian noise (AWGN) with a power spectral density of $N_0/2 = -60$dBm/Hz.

The project emphasizes the development of robust modulation and demodulation chains for AM and FM. In the amplitude modulation chain, specific attention is given to selecting transmission power, modulation type, and carrier frequency offset to ensure reliable communication. Similarly, the frequency modulation chain is constructed carefully considering these parameters to achieve efficient signal transmission and recovery.

A detailed analysis of the system's components is presented in the report, including justifications for bandwidth allocation, transmission power, and the modulation schemes employed. Simulink is utilized to simulate and analyze the system's performance, with the spectrum analyzer providing insights into the received signal power relative to the noise floor. Through this project, we aim to demonstrate the efficacy of analog communication techniques in handling multiplexed signals under noise conditions, providing a comprehensive understanding of system design and implementation.

1 • Amplitude Modulation & Demodulation (AM)

AM is a fundamental technique in analog communication systems, widely used for transmitting information over radio frequencies. In AM, the amplitude of a high-frequency carrier signal is varied in proportion to the instantaneous amplitude of the message signal, while the carrier frequency remains constant. This modulation scheme is particularly advantageous due to its simplicity in implementation and demodulation, making it suitable for various communication applications. However, the performance of AM systems can be influenced by noise, transmission power, and bandwidth constraints, necessitating careful design and optimization.

A detailed block diagram of the system design is provided to illustrate the implementation and performance of the AM communication system:

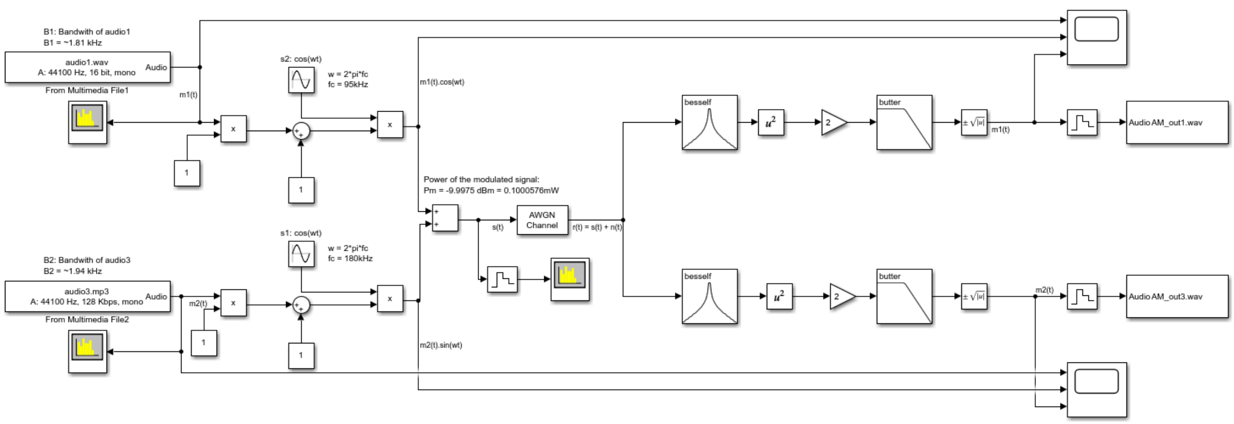


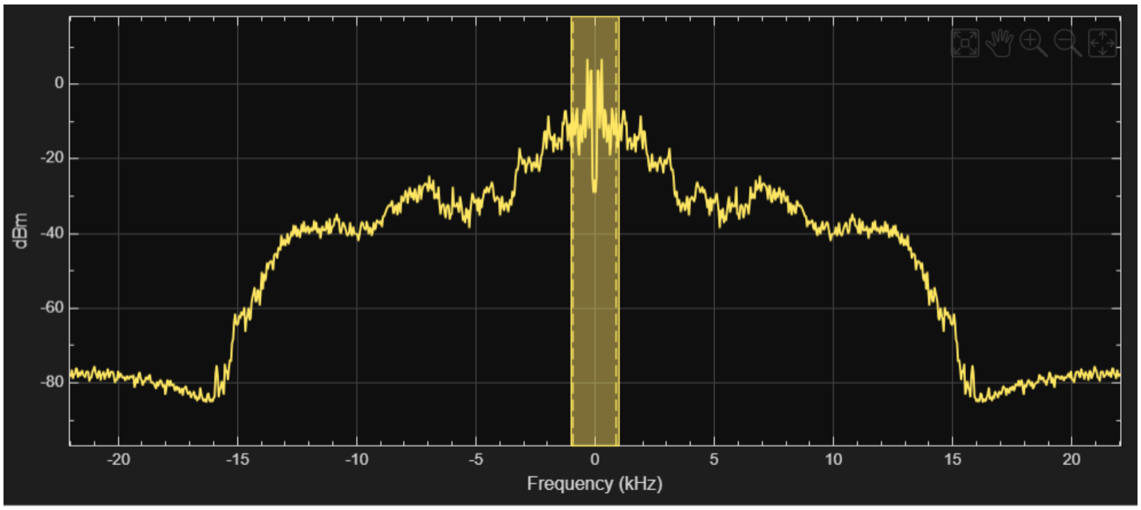
Figure 1: *Amplitude Modulation & Demodulation Block Diagram.*

The block diagram of the AM communication system consists of three primary stages: modulation, noise addition, and demodulation. During the modulation stage, two distinct audio signals are modulated independently using separate carrier frequencies and subsequently combined to form a composite signal. This composite signal is then transmitted through an additive white Gaussian noise (AWGN) channel, simulating realistic transmission environments.

In the demodulation stage, the received noisy signal is processed through bandpass filters centered at the respective carrier frequencies to separate the individual components. Envelope detection is then employed to demodulate the filtered signals, effectively recovering the original audio recordings. This systematic approach ensures reliable transmission and accurate recovery of the signals, even in the presence of noise.

1.1 ● Modulation Process

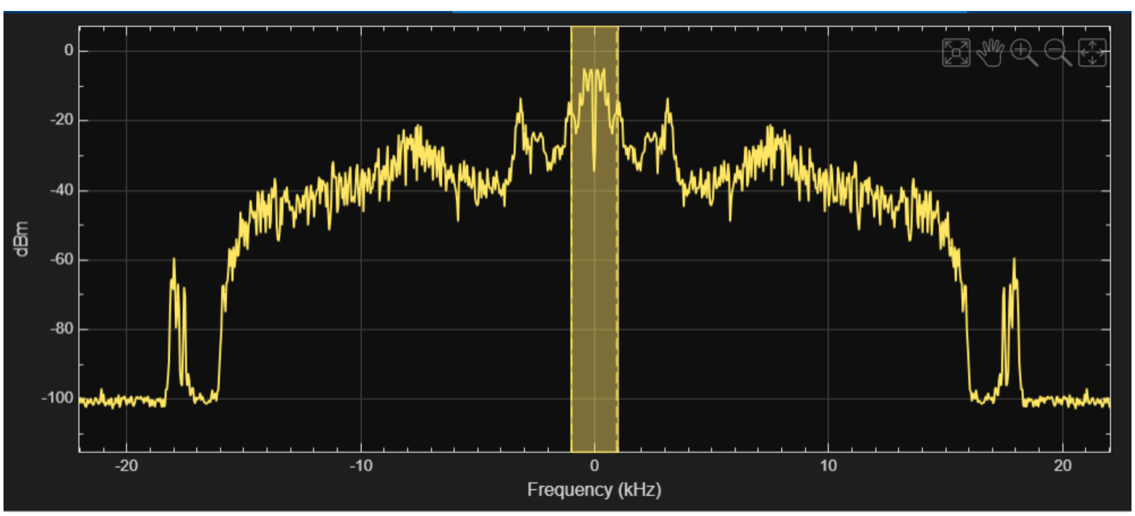
In the amplitude modulation process, two audio recordings, denoted as *audio1* and *audio3*, were first analyzed to determine their frequency characteristics. Both recordings have a frequency of 44100 Hz, with *audio1* exhibiting a bandwidth B_1 of approximately 1.81 kHz, and *audio3* having a bandwidth B_2 of around 1.94 kHz, as observed through the spectrum analyzer.



Occupied Bandwidth

m1(t)	Channel Power (dBm)	Occupied Bandwidth (kHz)	Frequency Error (fHz)
	13.2865	1.8143	-227.3737

Figure 2: *Signal $m_1(t)$ (audio1) in Frequency Domain with Transmission Power and Occupied Bandwidth.*



Occupied Bandwidth

m2(t)	Channel Power (dBm)	Occupied Bandwidth (kHz)	Frequency Error (fHz)
	6.2393	1.9382	170.5303

Figure 3: *Signal $m_2(t)$ (audio3) in Frequency Domain with Transmission Power and Occupied Bandwith.*

At the beginning of the modulation, multiplication by 1 is a placeholder for scaling the signal. Adding 1 is essential in amplitude modulation (AM) to shift the message signal upward. This ensures the signal remains non-negative, allowing the modulated signal's envelope to represent the original message accurately. This step is crucial for proper modulation and reliable demodulation, particularly when using envelope detectors, as it prevents distortion caused by negative portions of the message signal.

After this step, each audio signal was multiplied by a cosine wave, where the first audio signal m_1(t) (*audio1*) was modulated using a carrier frequency f_{c1} of 95 kHz, and the second audio signal m_2(t) (*audio3*) was modulated using a carrier frequency f_{c2} of 180 kHz. This multiplication process encoded the audio information onto the carrier waves.

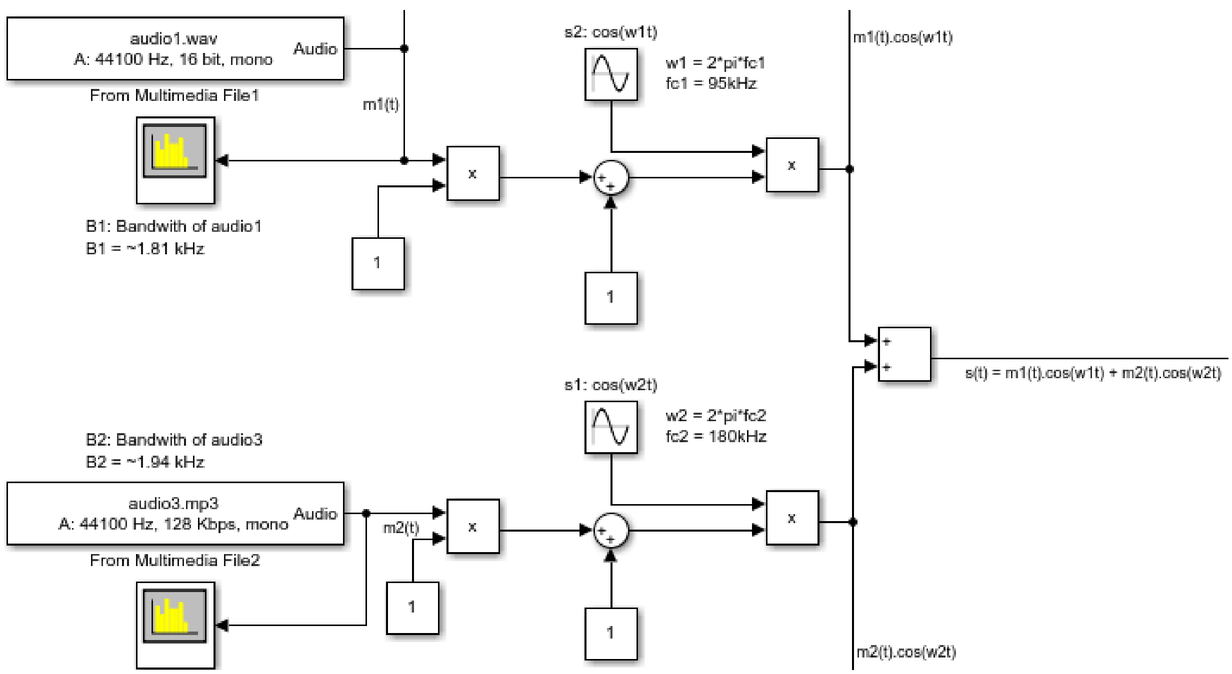


Figure 4: *AM Modulation Block Diagram (s(t): Modulated Signal).*

1.2 ● AWGN and Signal Power

The modulated signals' power before the noise addition was calculated using the spectrum analyzer. The combined power of the two modulated signals was determined to be $Pm = -9.9975$ dBm = 0.0561643 mW, which was subsequently used as an input value for the calculation of the signal-to-noise ratio (SNR) in the AWGN process.

This value was critical for adjusting the noise level to ensure the demodulation could recover the original signals accurately under noisy conditions.

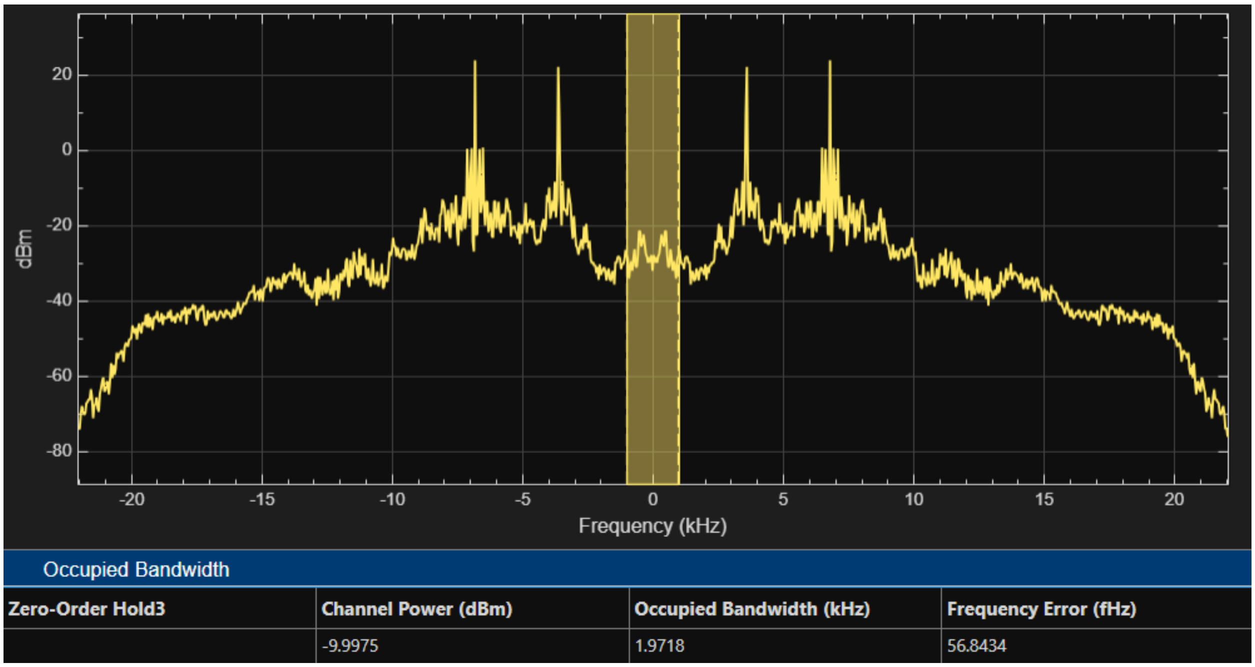


Figure 5: *Modulated Signal s(t) in Frequency Domain with Transmission Power and Occupied Bandwith.*

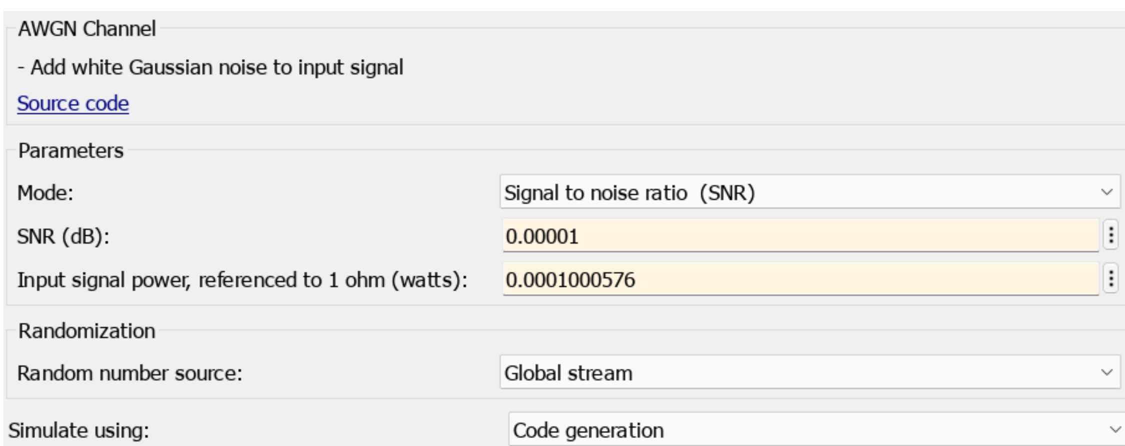


Figure 6: *Block Parameters of AWGN Channel.*

In the AWGN channel, the received signal $r(t)$ is modeled as the sum of the modulated signal $s(t)$ and the noise $n(t)$, such that $r(t) = s(t) + n(t)$. The noise $n(t)$ introduces random variations to the modulated signal, simulating real-world transmission conditions.

After passing through the AWGN channel block, the modulated signal is degraded by the added noise, which affects its clarity and makes it more challenging to recover the original information.

1.3 ● Demodulation Process

The purpose of the demodulation stage in AM communication is to recover the original message signals from the received modulated signal. This stage ensures that the transmitted information is accurately extracted despite the presence of noise.

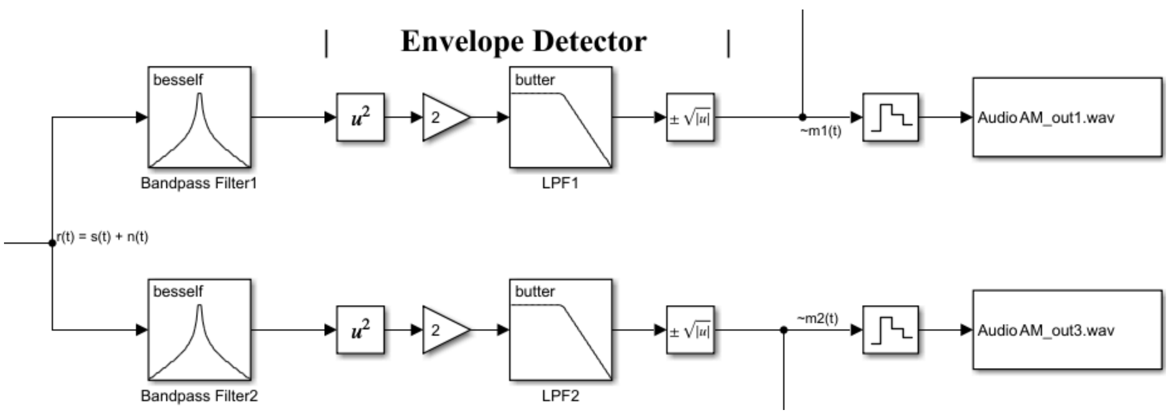


Figure 7: *AM Demodulation Block Diagram (r(t): Received Signal).*

The demodulation process was initiated after the modulated signals passed through the AWGN channel. The received signal *r(t)*, consisting of the modulated signal *s(t)* and noise *n(t)*, was first processed through bandpass filters centered around the carrier frequencies of 95 kHz and 180 kHz. These filters separated the composite signal into its individual components, isolating the two modulated signals corresponding to the original recordings.

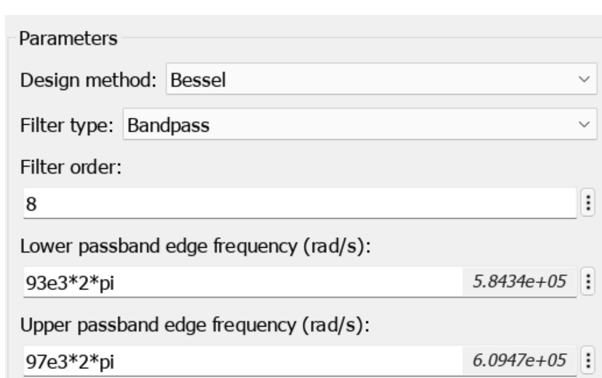


Figure 8: *Block Parameters of Bandpass Filter*$_1$*: Passband Edges =* $f_{c1} \pm B_1$.

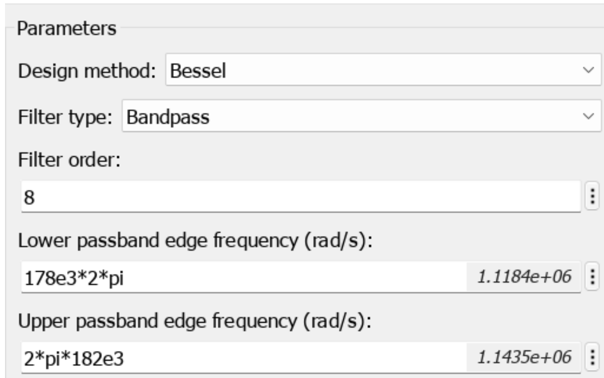


Figure 9: *Block Parameters of Bandpass Filter*$_2$*: Passband Edges =* $f_{c2} \pm B_2$.

Following the separation of the modulated signals, envelope detection was employed to recover the original audio signals by extracting the envelope of the modulated carrier wave, which encodes the transmitted information. The envelope detector consists of four components: a squaring block to amplify and rectify the signal, a gain block to scale the output, a low-pass filter (LPF) to remove high-frequency noise, and a square-root block to restore the original amplitude variations.

The LPFs, with passband edge frequencies of 44100 Hz, are aligned with the sampling rate of the input audio to ensure proper recovery of the message signal while filtering out residual noise and carrier components, enabling accurate extraction and recovery of the transmitted audio signals.

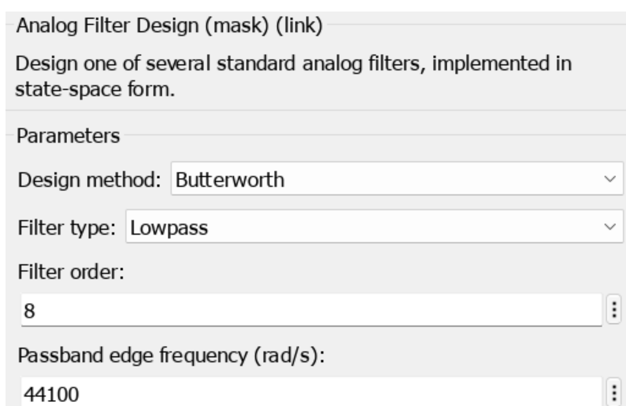


Figure 10: *Block Parameters of Lowpass Filters (*LPF_1 *&* LPF_2*).*

Finally, the outputs of the demodulation process were evaluated and compared to the initial audio recordings. The recovered signals were found to match the input audio files closely, demonstrating the success of the AM communication system in reliably transmitting and receiving voice messages despite the presence of noise.

2 • Frequency Modulation & Demodulation (FM)

Frequency Modulation (FM) is a technique in analog communication where the instantaneous frequency of a carrier wave is varied in proportion to the amplitude of the input message signal. This modulation method offers superior resistance to noise and interference, making it highly effective for transmitting high-quality audio signals. The primary purpose of FM is to encode information onto a carrier wave for reliable and efficient transmission over communication channels.

The architecture of the implemented FM system, as shown in the block diagram, is designed to modulate and demodulate two audio signals simultaneously:

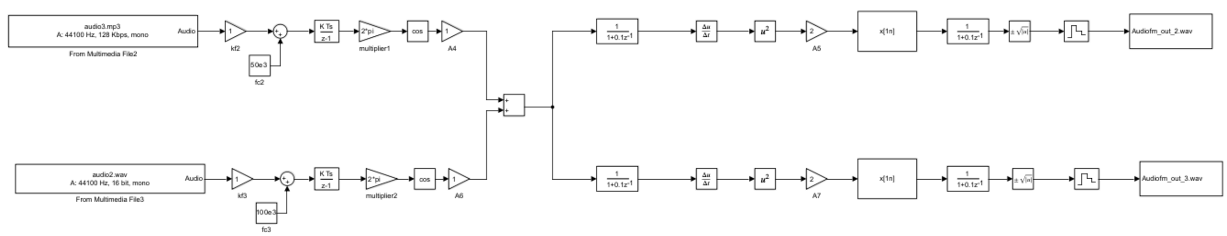


Figure 11: *Frequency Modulation & Demodulation Block Diagram.*

The FM communication system involves two main stages: modulation and demodulation. In the modulation process, two audio signals are independently modulated onto carrier frequencies of f_{c2} = 50 kHz, and f_{c3} = 100 kHz using frequency modulation, where f_i, the instantaneous frequency of each carrier, is varied in proportion to the message signal. The modulated signals are then combined into a composite signal for transmission.

In the demodulation stage, the composite signal is first separated into its individual components using discrete filters centered around the carrier frequencies. The separated signals are then passed through differentiators to convert frequency variations into amplitude variations. Finally, envelope detectors are used to extract the original audio signals, ensuring successful recovery of the transmitted information.

2.1 ● Modulation Process

The FM modulation process begins with the input audio signals. The first step involves taking the two audio signals and multiplying them by scaling factors, represented by the k_f block, to adjust their frequency deviation sensitivity, where k_f is proportionality constant known as the frequency sensitivity of the modulator. Each signal is processed separately using a series of blocks designed to modulate the signals onto their respective carrier frequencies: the first signal is modulated with a carrier at f_{c2} = 50 kHz and the second with a carrier at f_{c3} = 100 kHz. The instantaneous frequencies of modulated signals will be:

- $f_{i2} = f_{c2} + k_{f2}.m_2(t)$, $m_2(t)$: *audio3*
- $f_{i3} = f_{c3} + k_{f3}.m_3(t)$, $m_3(t)$: *audio2*

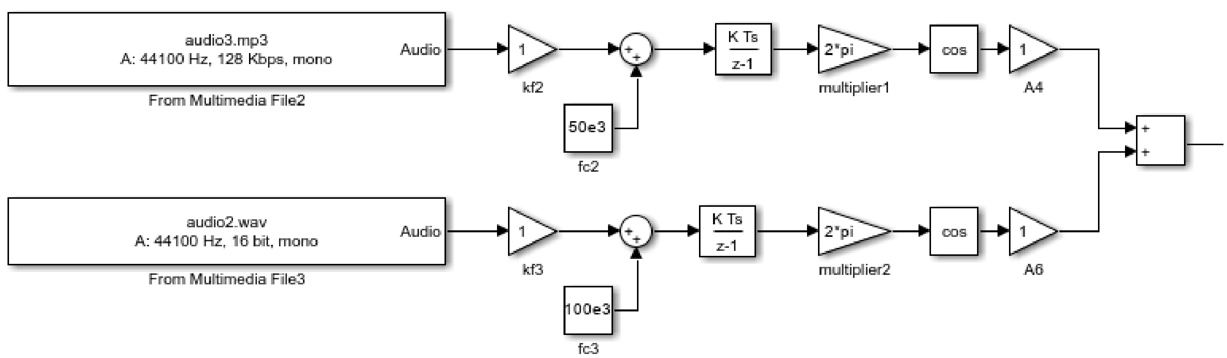


Figure 12: *FM Modulation Block Diagram.*

The signals are then passed through a discrete integrator block, which integrates the signals to produce the phase variation required for frequency modulation. This is followed by a gain block scaled by 2π, which helps to adjust the phase before passing the signal to a cosine block. The cosine block then generates the FM-modulated signal by applying the cosine function to the integrated input, creating a frequency-modulated waveform.

<u>Theoretical explanation:</u>

- After FM Modulation: $FM(t) = A_c \cos\theta_i$, where θ_i is the instantaneous carrier angle.
- Since $d\theta_i = 2\pi f_i dt$, $d\theta i = 2\pi f_i dt = 2\pi\{f_c + k_f\, m(t)\}dt \ = 2\pi f_c dt + 2\pi k_f m(t)dt$
- $\int d\theta i = \int\{2\pi f_c dt + 2\pi k_f m(t)dt\} \rightarrow \theta_i = \int 2\pi f_c dt + \int 2\pi k_f m(t)dt = 2\pi f_c\, t + 2\pi k_f \int m(t)dt$
- Finally: $FM(t) = A_c \cos\theta_i = A_c \cos[2\pi f_c\, t + k_f \int m(t)dt]$

After the modulation of both signals, the outputs are combined, creating a composite FM signal that contains both audio messages. The use of these blocks ensures that the two audio signals are effectively modulated onto their respective carriers, allowing them to be transmitted simultaneously in the FM spectrum.

2.2 ● Demodulation Process

The purpose of FM demodulation is to recover the original message signals encoded in the frequency variations of the FM-modulated carrier wave. Unlike amplitude modulation, where the information is contained in the signal's amplitude, FM requires extracting information from variations in the instantaneous frequency of the carrier.

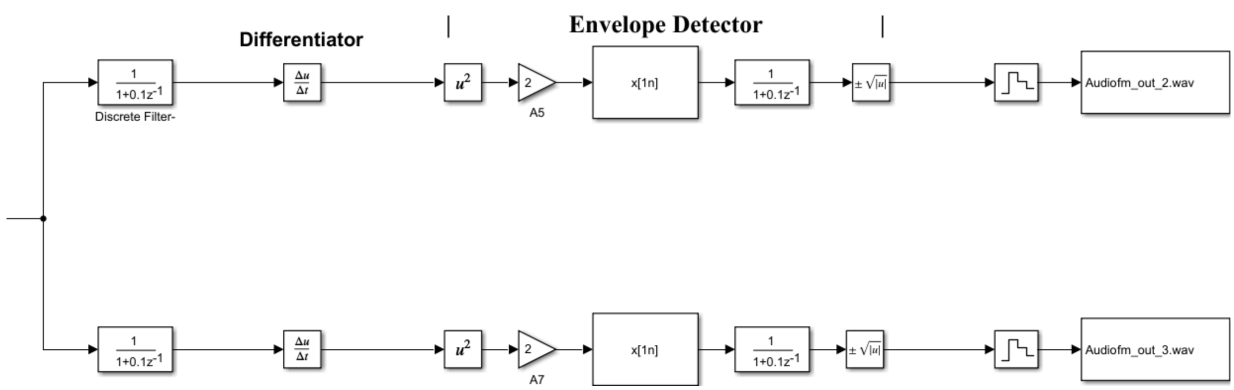


Figure 13: *FM Demodulation Block Diagram.*

The first discrete filter is employed to separate the two modulated signals, which were combined before demodulation. These filters are designed to isolate the frequency components centered at f_{c2} = 50 kHz and f_{c3} = 100 kHz, corresponding to $m_2(t)$ and $m_3(t)$, respectively. In practice, discrete filters are implemented using difference equations, which rely on the filter's transfer function in the z-domain. The filter's coefficients are determined by applying the z-transform to the desired frequency response, allowing precise separation of the signals.

The differentiator computes the time derivative of the filtered signal, converting the frequency variations of the modulated signal into amplitude variations. This is a crucial step in FM demodulation, as it translates the instantaneous frequency changes into a format that can be further processed to retrieve the message signals.

Theoretical explanation:

- $x(t) = dFM(t)/dt$ where $x(t)$ is the differentiated signal, $FM(t)$ is the modulated signal.
- $x(t) = \frac{d}{dt}(A_c \cos\theta_i) = -\frac{d}{dt}\theta_i(t) A_c \sin(\theta_i(t)) = 2\pi A_c f_i(t) \sin(2\pi f_c t + 2\pi k_f m(t)dt + \pi)$

  $= 2\pi A_c(f_c + k_f m(t)) \sin(2\pi f_c t + 2\pi k_f m(t)dt + \pi)$

After differentiation, the signal is passed through the envelope detector to extract the amplitude envelope, which corresponds to the original audio signals.

- Output of envelope detector: $y(t) = 2A_c(f_c + k_f m(t)) = 2A_c f_c + 2A_c k_f m(t)$
- Output of DC block: $z(t) = 2A_c k_f m(t) = Km(t)$

In the final stage of the FM demodulation process, the signal is passed through a zero-order hold block to ensure that both the input and output remain in digital form, maintaining compatibility with the system's discrete nature. The final block then converts the digitally demodulated signals into audible output, enabling a direct comparison with the original audio recordings to assess the accuracy of the transmission and recovery process.

References

1. Digital Introduction to Analog and Digital Communications, S. Haykin and M. Moher, John Wiley& Sons, 2010.

2. Modern Digital and Analog Communication Systems, B. P. Lathi Oxford Press, 2010.