Modeling and Performance Analysis of PCM Multiplexing and De-Multiplexing in MATLAB Simulink

Md. Mayn Uddin

Dept. of Electrical and Electronic Engineering, Jatiya Kabi Kazi Nazrul Islam University, Trishal, Mymensingh, Bangladesh

---***---

Abstract – *In the contemporary landscape of tele-communications, the paramount objective is the preservation of signal integrity. Effective modulation is indispensable for mitigating distortion and ensuring high-fidelity data transmission. This paper details the architectural design and implementation of a Pulse Code Modulation (PCM) system utilizing the MATLAB Simulink environment. PCM facilitates the digitization of diverse analog data, effectively streamlining systemic complexity. Furthermore, we explore Pulse Code Modulation Multiplexing (PCMM) to enable the concurrent transmission of multiple analog signals over a singular channel. Our methodology involves routing information signals through a quantizer and subsequently recovering them via demultiplexing and Low Pass Filtering (LPF). To validate signal characteristics, we conducted Auto-correlation and Cross-correlation analyses.*

***Key Words*: PCMM, Multiplexing techniques, PCM with MATLAB, PCMM simulation.**

1. INTRODUCTION

While baseband signals may be propagated directly over specific physical media, such as conventional telephony or facsimile lines, they frequently encounter spectral incompatibilities with the transmission channel. The intrinsic frequency-domain characteristics of many information sources are often ill-suited for direct transmission due to channel-induced attenuation or bandwidth constraints. Consequently, to ensure optimal signal propagation, message signals must undergo frequency translation to occupy the designated spectral bandwidth. This fundamental transformation, known as modulation, involves mapping the baseband signal onto a high-frequency carrier wave. To ensure the integrity of the data at the terminal point, the modulated signal must undergo a reciprocal operation—demodulation to achieve high-fidelity signal reconstruction [1]. Pulse Code Modulation (PCM) is the primary method for converting analog waveforms into discrete digital signals. In this framework, analog samples are represented as binary data, allowing multiple signals to be multiplexed and transmitted simultaneously without mutual interference. The process involves sampling the analog amplitude at uniform intervals and quantizing each measurement to the nearest discrete digital value. The fidelity of the resulting stream depends on two factors: the sampling rate, which defines the frequency of acquisition, and the bit depth, which determines the resolution of each sample [2],[3]. Modern PCM is driven by the refinement of sampling techniques and the translation of signals into sophisticated codes within practical hardware constraints. A compelling feature of this technology is the ability to strategically trade bandwidth for an enhanced signal-to-noise ratio, providing a distinct advantage in high-performance transmission. Furthermore, the integration of regenerative repeaters ensures that signal integrity remains remarkably consistent, with the signal-to-noise ratio staying largely independent of the total number of repeaters in the link [4],[5].

2. CODE MODULATION MULTIBLEXING AND DEMULTIPLEXING MODEL

There are three steps in the development of a PCM signal from that analog model:

1. Sampling;
2. Quantization, and
3. Coding.

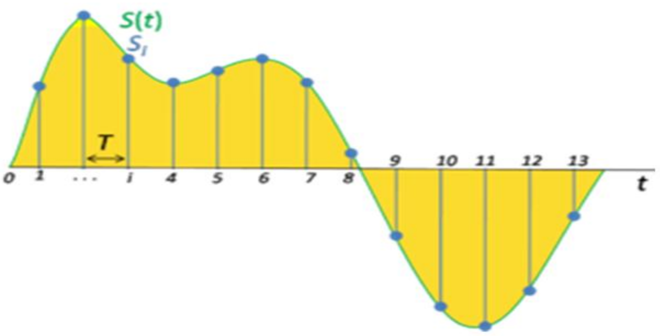

Figure 1: Sampling Process

In signal processing, sampling involves the transformation of a continuous-time waveform into a discrete-time sequence. A ubiquitous application of this principle is the digitization of acoustic waves into a series of discrete data points [6]. Technically, a sample represents the instantaneous magnitude of a signal at a specific temporal or spatial coordinate. The sampling frequency quantified in Hertz (Hz) or kilohertz (kHz) denotes the number of discrete observations captured per second; for instance, a rate of 44,100 samples per second is designated as 44.1 kHz. Furthermore, bandwidth is defined as the spectral range between the maximum and minimum frequency components contained within a specific transmission stream [4].

The input message signal is interfaced with a sequence of narrow rectangular pulses to approximate the instantaneous sampling process. To facilitate high-fidelity reconstruction at the receiver, the sampling frequency must exceed the Nyquist rate, which is double the maximum frequency component W of the message signal. In practical implementations, a low-pass anti-aliasing filter is integrated at the pre-sampling stage to attenuate frequency components exceeding W, thereby preventing spectral overlap. Consequently, the sampling procedure enables the conversion of a continuous waveform into a finite set of discrete-time values[6],[7]. In the domains of mathematics and digital signal processing, quantization represents the systematic mapping of continuous input amplitudes onto a discrete, finite set of output levels. This procedure essentially transforms an infinite range of values into a manageable, countable distribution. Fundamental techniques such as rounding and truncation serve as primary examples of this reductive process, where signal precision is traded for digital representability [8],[9].

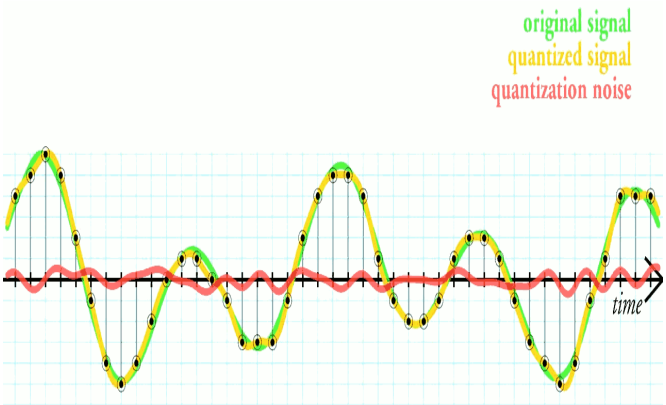

Figure 2: Quantization process

In the context of digital signal processing, several inherent limitations exist within the uniform quantization framework. Specifically, uniform quantization is only mathematically optimal for signals with a uniform distribution, whereas authentic audio signals exhibit a high concentration of energy near the zero-axis. Furthermore, the psychoacoustic properties of human hearing render us significantly more sensitive to quantization distortions at lower amplitudes. To mitigate these discrepancies, non-uniform quantization is employed to provide a more nuanced resolution for near-zero values [10]. In instances where non-uniform quantization is not feasible, an alternative approach involves the utilization of adaptive quantization or differential encoding. Rather than maintaining a static distribution of levels, these systems dynamically adjust the step size based on the variance of the input signal, ensuring the quantizer remains optimized for the signal's instantaneous dynamic range. Furthermore, if the source signal is truly random or possesses a flat probability density function, uniform quantization remains the most computationally efficient choice. However, for most physical phenomena such as speech or biological data the shift toward non-uniform or adaptive methodologies is essential to minimize the mean squared error (MSE) and maximize the Signal-to-Quantization-Noise Ratio (SQNR) [11].

In telecommunications, a code is a systematic framework used to convert information such as text, audio, or imagery into an alternative format for efficient transmission or storage. While the advent of language allowed for the articulation of thought, it was geographically and temporally constrained. The subsequent development of writing transfigured oral discourse into visual symbols, effectively extending the reach of communication across vast distances and historical epochs. Encoding facilitates the transformation of source information into a symbolic format optimized for transmission or archival, while decoding serves as the reciprocal process, restoring those symbols into an intelligible state for the recipient. While the integration of sampling and quantization constrains a continuous message to a discrete set of values, this format remains suboptimal for direct propagation over telephonic or radio frequencies. To leverage the inherent benefits of these processes specifically to bolster signal resilience against noise, interference, and channel degradation an encoding procedure is required to transfigure discrete samples into a robust signaling format[12]. In this context, a code is defined as a systematic arrangement of discrete events, or symbols, used to represent each value. A singular instance within this set is termed a code element; for example, the binary state of a pulse (its presence or absence) constitutes a symbol. A unique configuration of these symbols designed to represent a specific discrete value is designated as a code word or character. In a binary framework, each symbol is restricted to two distinct states, such as the presence or absence of a pulse, typically denoted by the bits 0 and 1. While ternary codes utilize three discrete values, binary encoding remains the superior choice for mitigating channel noise. This is because binary symbols possess a high degree of noise immunity and are easily reconstructed through regenerative repeaters. In a code where each word consists of R bits (the acronym for binary digits), a total of 2^R unique values can be represented. For instance, an 8-bit code word can effectively map 256 discrete quantization levels. A standard methodology for establishing a one-to-one correspondence between these levels and their respective code words is to utilize the ordinal number of the level expressed in base 2. In this binary system, each digit's significance is determined by its positional weight as a power of 2[13] .

Digital communication systems utilize various line codes to translate binary data into electrical signals, each offering unique trade-offs in power efficiency and spectral density. Unipolar Non-Return-to-Zero (NRZ), or on-off signaling, represents a binary '1' with amplitude A and '0' by switching the pulse off. While straightforward, it is inefficient due to a persistent DC offset and poor spectral characteristics at low frequencies. In contrast, Polar NRZ employs symmetric pulses of +A and -A, which is simple to implement but still suffers from significant power concentration near zero frequency. To address synchronization needs, Unipolar Return-to-Zero (RZ) uses half-width pulses for binary '1', creating spectral delta functions that aid in bit-timing recovery at the receiver; however, this comes at the cost of a 3 dB power penalty compared to polar schemes. Bipolar Return-to-Zero (BRZ), also known as Alternate Mark Inversion (AMI), utilizes three levels where binary '1's alternate between +A and -A. This effectively eliminates the DC component and minimizes low-frequency interference. Finally, Split-Phase (Manchester) coding ensures a transition in the center of every bit interval a positive-to-negative swing for '1' and the reverse for '0'. This technique provides superior DC suppression and self-clocking capabilities, making it highly robust for complex transmission environments [14]. In a Pulse Code Modulation (PCM) system, multiplexing is integrated directly into the sampling phase by sequentially capturing data from various analog sources. These sources, typically nominal 4-kHz voice channels or bandwidth-limited data such as freeze-frame video, are interleaved in time. The resulting stream of sampled and quantized values forms a serial bitstream of 1s and 0s, which serves as the digital translation of the original signal levels. To ensure the coherent reconstruction of this data at the destination, the serial bitstream requires a specific synchronization mechanism. Because the receiver must distinguish where a sampling sequence begins, an identification marker known as a framing bit is introduced. In a standard DS1 configuration, once the receiver detects this indication, it is programmed a priori to expect a specific sequence (such as 24 eight-bit slots) to follow. This complete cycle of samples, including the synchronization overhead, is formally designated as a frame. In digital communications, a single transmission line is often tasked with carrying multiple signals. However, since only one signal can occupy a physical line at any given instant, a specialized device is required to manage access. This device is the multiplexer, commonly abbreviated as a Mux. The primary function of a multiplexer is to select one input from n available lines and route it to a single output. This allows for efficient resource sharing and data compression across a common transmission medium. Conversely, the de-multiplexer (De-Mux) performs the inverse operation, taking the single signal from the shared line and redistributing it to the appropriate output destination. In complex large-scale systems, certain integrated circuits are capable of performing both multiplexing and de-multiplexing operations simultaneously[15].

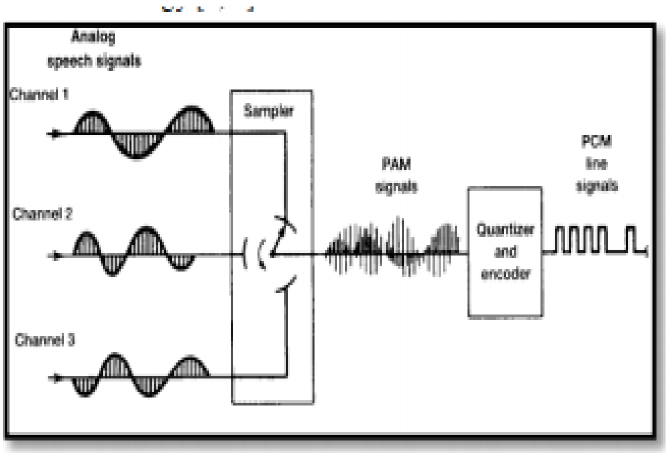

Figure 3: pulse code modulation multiplexer (PCMM)

In digital communications, a multiplexer (Mux) acts as a high-speed electronic data selector, enabling multiple information streams to share a single transmission link simultaneously. While a single-pole multi-position switch provides a simple mechanical analogy for this process, modern systems utilize electronic components to achieve the rapid switching necessary for complex signals. At the destination, the receiver employs a demultiplexer (De-Mux) to perform the inverse operation, separating the combined signal back into its original, independent channels. This coordinated process of "Muxing" and "De-Muxing" is essential for maximizing the efficiency of communication channels and allowing for high-capacity data transmission [16].

Multiplexers are versatile components capable of managing both analog and digital signals. In analog applications, they typically employ physical relays or transistor switches to route continuous waveforms. Conversely, in digital applications, they are constructed from standard logic gates and are formally categorized as digital multiplexers. Complementing this is the de-multiplexer, a device engineered with a single input and multiple output lines, used to dispatch a signal to one of several specific destinations. The fundamental distinction lies in their operational direction: a multiplexer consolidates two or more signals to encode them onto a shared transmission medium, while the de-multiplexer executes the inverse operation, decoding the shared stream back into its constituent signals [17],[18].

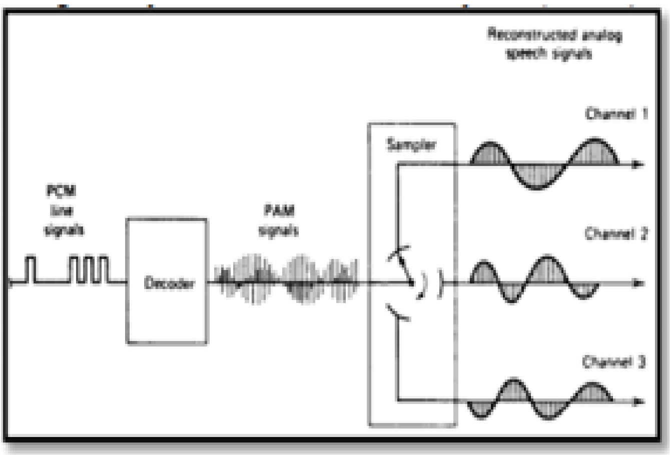

Figure 4: pulse code modulation De-multiplexer (PCMM

Multiplexers and de-multiplexers are the foundational "traffic controllers" of modern digital infrastructure. By allowing multiple data streams to share a single physical line, multiplexers maximize channel efficiency in applications ranging from telephone networks to satellite telemetry. This consolidation is not just a matter of convenience; it is essential for managing massive amounts of computer memory and high-capacity fiber-optic links where individual wiring for every signal would be physically impossible. At the receiving end, the de-multiplexer performs the vital task of redistribution, acting as a serial-to-parallel converter that routes incoming data to the correct destination, such as specific CPU registers or user devices. This entire process relies on precise electronic switching and synchronization—ensuring that every packet of audio, video, or data is extracted and delivered with total integrity [19],[20].

Table 1

Quantizer level	CODE	Quantizer level	CODE
0	0000	8	1000
1	0001	9	1001
2	0010	10	1010
3	0011	11	1011
4	0100	12	1100
5	0101	13	1101
6	0110	14	1110
7	0111	15	1111

3. PULSE CODE MODULATION MULTIPLEXING SIMULATION

Pulse code modulation multiplexing transmitter is illustrated in figure 5.

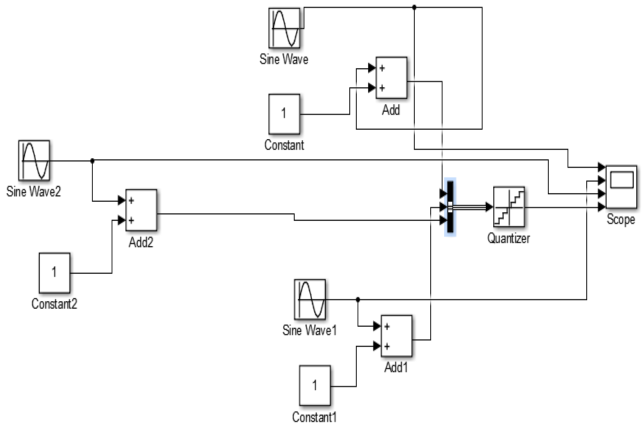

Figure 5: PCMM transmitter

In this simulation, set to a 5-second duration, the information signals undergo a DC shift to move their amplitudes entirely above the zero axis. This ensures the waveforms are unipolar before they are interleaved by the multiplexer and passed to the quantizer. The quantizer then maps these continuous-amplitude samples onto a finite set of discrete levels, effectively digitizing the signal's intensity. As seen in Figure 6, this transformation converts the raw analog input into a quantized format suitable for PCM transmission, illustrating the transition from a smooth wave to a stepped digital representation.

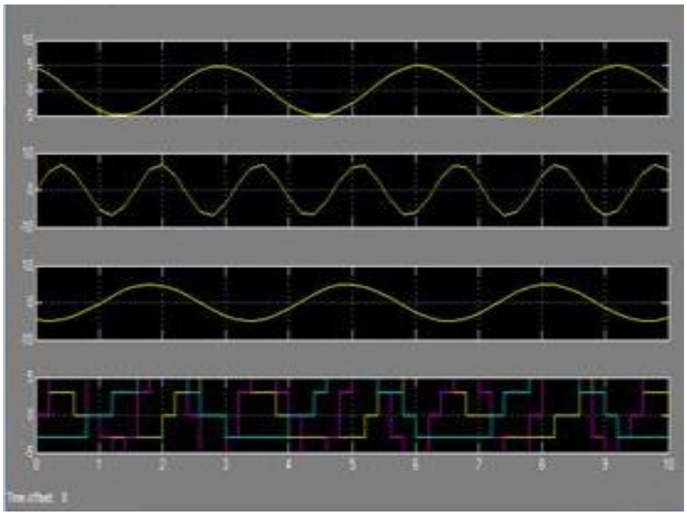

Figure 6: a. Information signal 1, b. Information signal 2, c. Information 3, 4-PCMM Signal.

The transition from a quantized signal to a Digital Pulse Code Modulation (D-PCM) signal involves an encoding process where each discrete amplitude level is translated into a specific serial binary code. In this specific configuration, the three multiplexed signals are assigned different bit depths, which directly influences the resolution and precision of their digital representation. Signal 1 is encoded using a 3-bit scheme, allowing for $2^3 = 8$ distinct quantization levels. Signal 2 utilizes a higher resolution 4-bit code, providing $2^4 = 16$ levels. Finally, Signal 3 is the most precise, employing a 5-bit conversion that results in $2^5 = 32$ discrete levels. This variation in bit allocation suggests a multi-rate or multi-resolution system where different channels require different levels of fidelity.

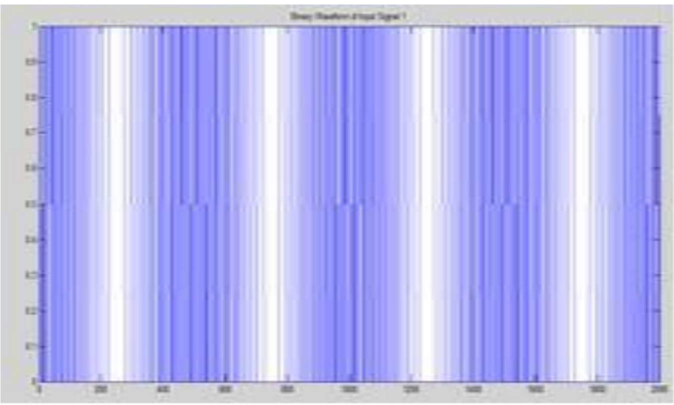

Figure 7: signal 1 binary waveform.

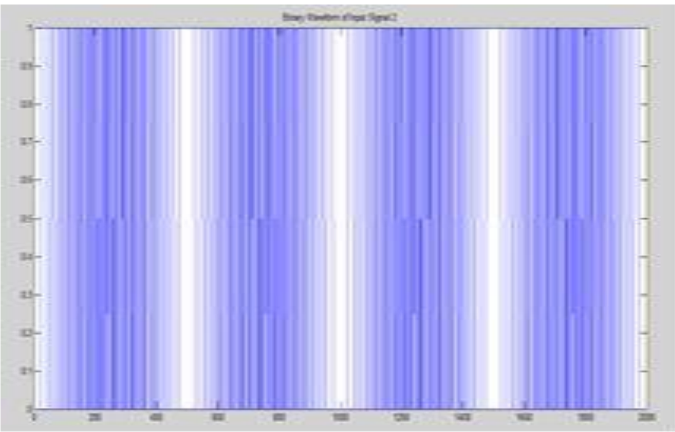

Figure 8: Signal 2 binary waveform

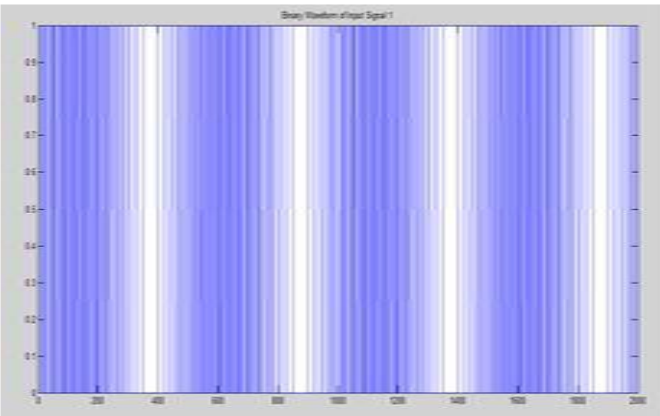

Figure 9: Signal 3 binary waveform

Autocorrelation is the mathematical tool used to measure the similarity between a signal and a delayed version of itself. By comparing a bitstream to a time-lagged copy, I can identify recurring patterns, periodicity, and the overall power distribution of the signal. In my test of binary signals, the main lobe shown in figure 10 represents the point of maximum correlation where the signal perfectly aligns with itself at zero lag. The characteristics of this lobe are vital for determining signal quality, as they reveal the bit duration and help predict how much the pulses might spread or overlap during transmission.

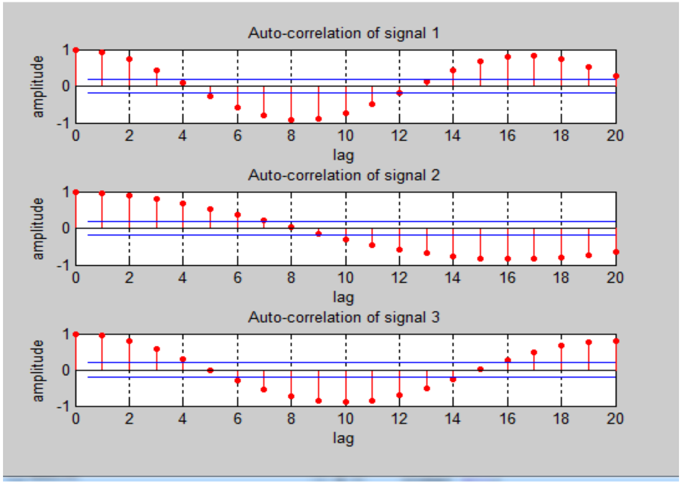

Figure 10: Autocorrelation test on signal 1, signal 2 and signal 3.

Cross-correlation serves as a mathematical measure of the similarity between two distinct signals as one is shifted in time relative to the other. Often described as a sliding dot product, this process involves moving one variable signal across another to identify points of alignment or shared patterns. While autocorrelation compares a signal to itself, cross-correlation identifies the relationship between different data streams within a system. In your specific implementation, the cross-correlation between binary signal 1, signal 2, and signal 3 as illustrated in figure 11 is used to evaluate the independence of the channels. By quantizing these input signals before multiplexing, the system effectively minimizes crosstalk. This occurs because quantization forces the signals into discrete, well-defined levels, making it easier for the receiver to distinguish between the intended data of one channel and the unwanted interference leaked from another.

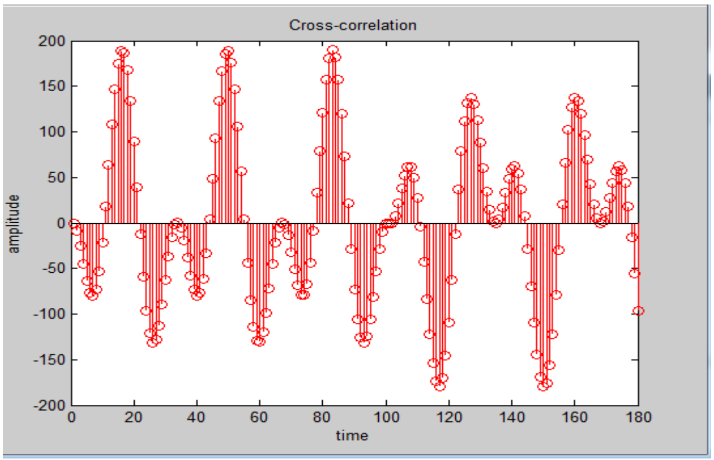

Figure 11: Cross correlation test on signal 1, signal 2 and signal 3

4. PULSE CODE MODULATION DE-MULTIPLEXING SIMULATION

In the final stage shown in figure 12, the de-multiplexer separates the serial bitstream back into individual channels. To restore the original analog waveforms, Low-Pass Filters (LPF) are applied to smooth out the discrete quantization steps and recover the baseband signal. Once filtered, the signals are re-shifted downward by their respective amplitudes (A1, A2, A3) to remove the DC bias added during transmission. Finally, an amplifier with gain K scales the recovered signals back to their original levels completing the reconstruction process

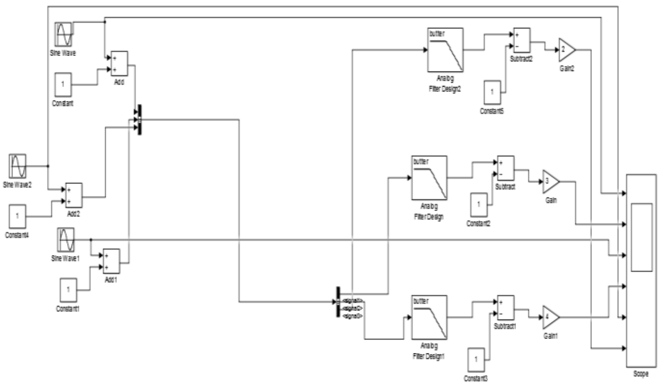

Figure 12: PCM De-Multiplexing

Waveforms at the scope can be shown in figure 13.

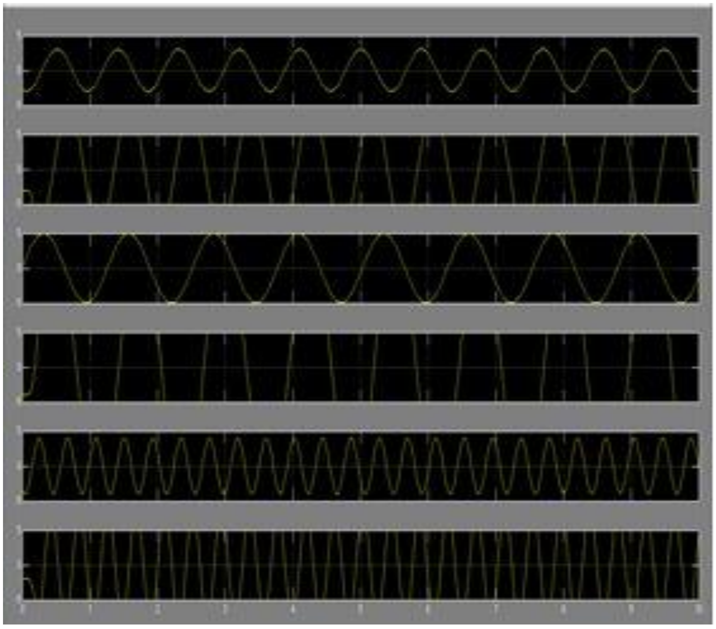

Figure 13: a. Signal 1, b. Recovered signal 1, c. Signal 2, d. Recovered signal 2, e. Signal 3, f. Recovered signal 3.

From this figure it could be shown that the recovered signal had a delay time. Delay time τ of signal 1 is 0.002s, signal 2 $\tau 2$ = 0.001s and signal 3 $\tau 3$ = 0.0001s. Since $f1 < f2 < f3$ so $\tau 1 > \tau 2 > \tau 3$. To overcome the delay time τ on the recovered signal, two methods are used.

1. Reduce the order of low pass filter
2. Using the delay block in the simulation model as shown in fig 12.

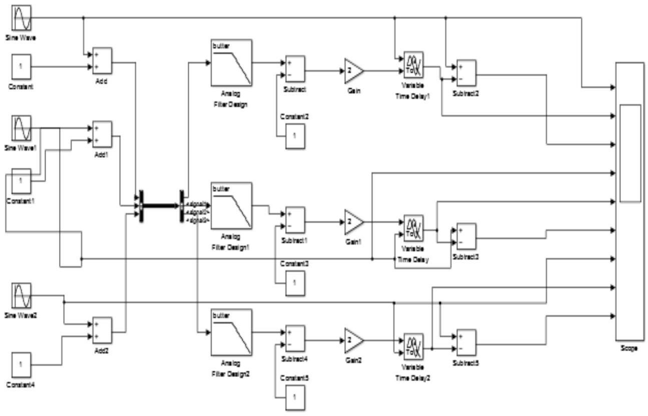

Figure 14: PCMM De-multiplexing with time delay cancelation

Output signals of figure 14 are shown in figure 15, where the delay time in recovered signals canceled.

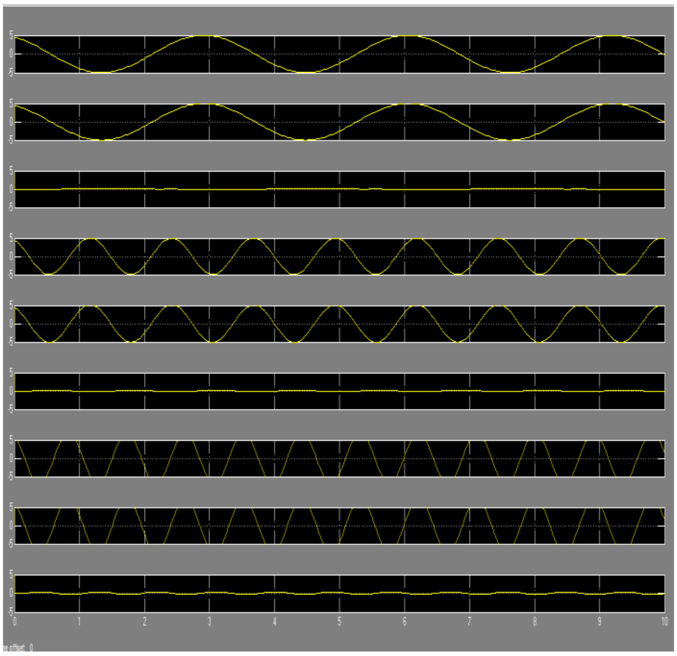

Figure 15: a. Signal 1, b. Recovered signal 1, c. delay of signal 1 d. Signal 2, e. Recovered signal2, f. delay of signal 2, g Signal 3, h. Recovered signal 3, i. delay of signal 3.

5. CONCLUSION

The implementation of Pulse Code Modulation (PCM) multiplexing and de-multiplexing in MATLAB Simulink successfully achieved high-performance benchmarks. The robustness of the design is validated by the autocorrelation and cross-correlation results, which confirm high signal inequality and the total absence of inter-channel interference. These metrics prove that each multiplexed signal remains distinct and identifiable, ensuring the integrity of the multi-input stream during transmission. At the receiver end, the common challenge of propagation and processing delay was effectively mitigated through strategic model optimization. By reducing the order of the Low-Pass Filters (LPF) and fine-tuning the delay blocks within the Simulink library, the phase shift was minimized. This adjustment allows the reconstructed output to align precisely with the original input signals in the time domain, resulting in a high-fidelity recovery process with negligible latency.

ACKNOWLEDGEMENT

The authors can acknowledge the use of generative AI tools such as ChatGPT, Grammarly only for making the grammatical corrections and increase the readability.

REFERENCES

[1] R. E. Blahut, Modem Theory: An Introduction to Telecommunications, 1st edn. Cambridge University Press, 2009. doi: 10.1017/CBO9780511811401.

[2] A. S. Holevo, ‘The capacity of the quantum channel with general signal states’, IEEE Trans. Inf. Theory, vol. 44, no. 1, pp. 269–273, Jan. 1998, doi: 10.1109/18.651037.

[3] G. Hughes, ‘On the mean accuracy of statistical pattern recognizers’, IEEE Trans. Inf. Theory, vol. 14, no. 1, pp. 55–63, Jan. 1968, doi: 10.1109/TIT.1968.1054102.

[4] C. E. Shannon, ‘Communication in the Presence of Noise’, Proc. IRE, vol. 37, no. 1, pp. 10–21, Jan. 1949, doi: 10.1109/JRPROC.1949.232969.

[5] C. E. Shannon, ‘A Mathematical Theory of Communication’, Bell Syst. Tech. J., vol. 27, no. 3, pp. 379–423, July 1948, doi: 10.1002/j.1538-7305.1948.tb01338.x.

[6] A. J. Jerri, ‘The Shannon sampling theorem—Its various extensions and applications: A tutorial review’, Proc. IEEE, vol. 65, no. 11, pp. 1565–1596, 1977, doi: 10.1109/PROC.1977.10771.

[7] J. Max, ‘Quantizing for minimum distortion’, IEEE Trans. Inf. Theory, vol. 6, no. 1, pp. 7–12, Mar. 1960, doi: 10.1109/TIT.1960.1057548.

[8] W. R. Bennett, ‘Spectra of Quantized Signals’, Bell Syst. Tech. J., vol. 27, no. 3, pp. 446–472, July 1948, doi: 10.1002/j.1538-7305.1948.tb01340.x.

[9] G. Rivard, ‘Direct fast Fourier transform of bivariate functions’, IEEE Trans. Acoust. Speech Signal Process., vol. 25, no. 3, pp. 250–252, June 1977, doi: 10.1109/TASSP.1977.1162951.

[10] E. Zwicker and H. Fastl, Psychoacoustics, vol. 22. in Springer Series in Information Sciences, vol. 22. Berlin, Heidelberg: Springer Berlin Heidelberg, 1999. doi: 10.1007/978-3-662-09562-1.

[11] J. Thomas, ‘Review of “Detection, Estimation, and Modulation Theory, Part III--Radar-Sonar Signal Processing and Gaussian Signals in Noise” (Van Trees, H. L.; 1971)’, IEEE Trans. Inf. Theory, vol. 18, no. 4, pp. 542–542, July 1972, doi: 10.1109/TIT.1972.1054844.

[12] R. W. Hamming, ‘Error Detecting and Error Correcting Codes’, Bell Syst. Tech. J., vol. 29, no. 2, pp. 147–160, Apr. 1950, doi: 10.1002/j.1538-7305.1950.tb00463.x.

[13] T. M. Cover and J. A. Thomas, Elements of Information Theory, 1st edn. Wiley, 2005. doi: 10.1002/047174882X.

[14] Md. S. Reza, Md. M. Hossain, A. A. Chowdhury, S. M. S. Reza, and Md. M. Rahman, 'Performance Evaluation of SCM-WDM System Using Different Linecoding', 2010, doi: 10.48550/ARXIV.1004.4830.

[15] S. Faruque, 'Time Division Multiplexing (TDM)', in Radio Frequency Source Coding Made Easy, in SpringerBriefs in Electrical and Computer Engineering. , Cham: Springer International Publishing, 2015, pp. 91–118. doi: 10.1007/978-3-319-15609-5_5.

[16] A. Grami, 'Communication Networks', in Introduction to Digital Communications, Elsevier, 2016, pp. 457–491. doi: 10.1016/B978-0-12-407682-2.00011-9.

[17] J. Ma, N. J. Karl, S. Bretin, G. Ducournau, and D. M. Mittleman, 'Frequency-division multiplexer and demultiplexer for terahertz wireless links', Nat. Commun., vol. 8, no. 1, p. 729, Sept. 2017, doi: 10.1038/s41467-017-00877-x.

[18] J. Haveman and J. Lavorel, 'Identification of the 120 mus phase in the decay of delayed fluorescence in spinach chloroplasts and subchloroplast particles as the intrinsic back reaction. The dependence of the level of this phase on the thylakoids internal pH', Biochim. Biophys. Acta, vol. 408, no. 3, pp. 269–238, Dec. 1975, doi: 10.1016/0005-2728(75)90129-2.

[19] A. Atieh, M. Waad, and I. Mansour, 'Effect of multiplexer/demultiplexer bandwidth on upgrading current 10G to 40 G optical communication systems', Jordanian J. Comput. Inf. Technol., vol. 3, no. 2, p. 131, 2017, doi: 10.5455/jjcit.71-1497643654.

[20] P. Kheirkhah Sangdeh and H. Zeng, 'Overview of Multiplexing Techniques in Wireless Networks', in Multiplexing, S. Mohammady, Ed., IntechOpen, 2019. doi: 10.5772/intechopen.85755.