International Journal of Reconfigurable and Embedded Systems (IJRES)
Vol. 15, No. 1, March 2026, pp. 86∼96
ISSN: 2089-4864, DOI: 10.11591/ijres.v15.i1.pp86-96

FPGA implementation and bit error rate analysis of the forward error correction algorithms in voice signals

Ramjan Khatik, Afzal Shaikh, Shraddha Sawant, Pritika Patil
Department of Electrical and Computer Engineering, School of Engineering and Technology, Anjuman-I-Islam's Kalsekar Technical Campus, New Panvel, India

Article Info

Article history:

Received Apr 1, 2025
Revised Jan 21, 2026
Accepted Jan 24, 2026

Keywords:

Convolution encoding
Field programmable gate arrays
Microblaze
Spread spectrum
Viterbi decoder

ABSTRACT

The idea of codes (VITERBI) is broadly utilized as a part of the wireless communication system as a result of their less complex nature in the decoding of transmitted message. This paper attempts to develop a performance analysis of the decoder by methods for bit error rate (BER) examination. The Galois field-based decoder calculation is only utilized as a part of the communication systems. The decoder calculation-based Viterbi based decoder is carried out using field programmable gate arrays (FPGA) and MATLAB. This paper looks at the execution examination of both the calculations. The reconfigurable processor called Microblaze on the Spartan 3E FPGA is utilized for this purpose. MATLAB based code is used to see the BER analysis after the FPGA implementation output.

Corresponding Author:

Afzal Shaikh
Department of Electrical and Computer Engineering, School of Engineering and Technology
Anjuman-I-Islam's Kalsekar Technical Campus
New Panvel, Navi Mumbai, India
Email: afzal.aiktc@gmail.com

1. INTRODUCTION

In his landmark 1948 work, Shannon demonstrated that every communication channel has a fundamental upper bound on the amount of information that can be transmitted reliably, determined by the level of noise present in the system [1]. Although this theoretical limit established the best possible performance of any coding strategy, it did not provide a constructive method for achieving it. Viterbi-based codes represent a family of block codes characterized by both long code lengths and randomized structure, and they are known to offer near-optimal performance, operating within approximately 0.0045 dB of the Shannon limit [2]. Among the various decoding strategies, message-passing techniques—commonly referred to as the sum–product or belief-propagation (BP) algorithms—are the most widely adopted in many implementations [3]-[8]. A fully parallel decoder performs the check-node update in one clock cycle and the bit-node update in the next, yielding a substantial increase in decoding throughput compared with serial architectures. However, a major limitation of these decoders is their lack of flexibility: existing parallel implementations [9]-[12] cannot easily accommodate modifications to code structure. Any change in code parameters, such as rate or degree distribution, typically requires redesigning the routing between check and bit nodes, making adaptation to new code configurations difficult.

The Spartan-3E field programmable gate array (FPGA) Kit board features the one of a kind highlights of the Spartan-3E FPGA family and gives an advantageous advancement board to embedded processing application. The Kit board features these highlights: the Micro blaze™ embedded processor soft core is a reduced

instruction set computer (RISC) advanced for usage in Xilinx® Field Programmable Gate Arrays (FPGAs). Figure 1 demonstrates an utilitarian piece outline of the Micro blaze center [13]-[16].

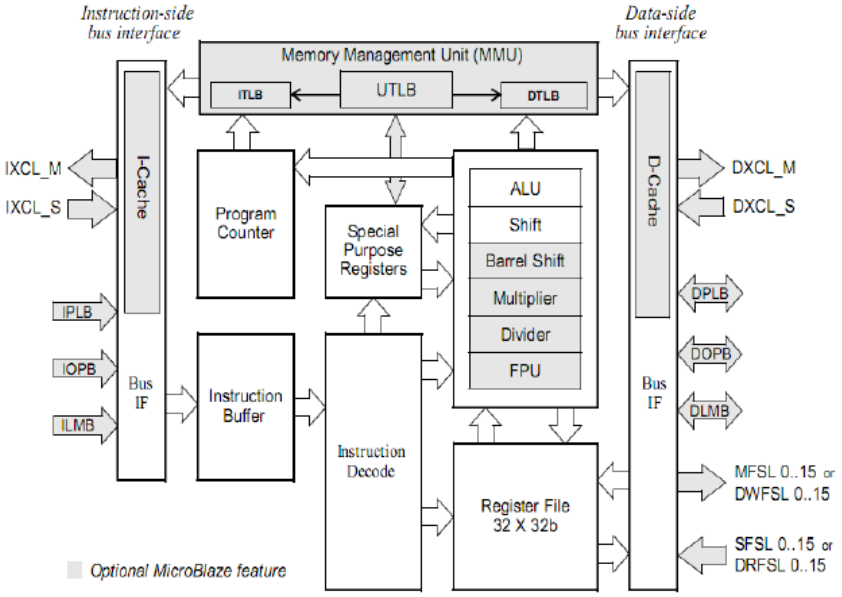

Figure 1. MicroBlze core block diagram [13]

Fully robotized age HDL design using the EDK Platgen/Simgen tools. Address range, byte-write enable count, information breadth, and target architecture are some of the design criteria that affect the number of BRAM primitives employed. The memory block's Ports A and B can be linked to the following free BRAM interface controllers: on-chip memory (OCM), processor local bus (PLB), on-chip peripheral bus (OPB), and local memory bus (LMB). As long as the appropriate amount of byte-write enables are set up, it supports byte, half-word, and double word exchanges [17]-[20].

2. METHOD

In this implementation, the voice input is initially captured and transformed into text-based data within MATLAB, as presented in Figure 2. Two voice samples from different channels are processed independently. After interpolation with the spreading sequence, the spread signal undergoes convolutional encoding as described in the previous sections. The encoded symbols are then modulated using quadrature phase shift keying (QPSK), followed by the addition of additive white Gaussian noise (AWGN) to simulate channel impairments. QPSK demodulation is performed to retrieve the noisy transmitted data. This processing pipeline is executed for a multichannel code division multiple access (CDMA) system, with all stages up to this point implemented in MATLAB. The demodulated output is subsequently stored in text format for flexibility in further programmatic manipulation. Error correction is achieved through the Viterbi algorithm, whose output—representing the reconstructed information—is also saved as a text file. Finally, the recovered signal is plotted in MATLAB and compared with the original waveform to assess reconstruction accuracy [21]-[27].

2.1. Flow chart explanation

In the design workflow, the Xilinx Platform Studio integrated within Xilinx ISE 10.1 employs header files generated from voice samples—such as those used in mobile CDMA systems—or from other signal sources processed through MATLAB's signal processing environment. Embedded C is utilized to implement the Viterbi encoding procedure, with dedicated routines developed for both the encoding and decoding operations. The signal sampled in MATLAB is converted into a header file and subsequently processed through the Viterbi encoder, after which QPSK modulation is applied. To emulate wireless transmission conditions, additive noise is introduced into the modulated data stream. The corrupted signal is then subjected to QPSK demodulation, followed by Viterbi decoding to recover the transmitted information. Post-decoding, error de-

tection and correction mechanisms are executed. During the iterative error-correction stage, the system's performance is assessed through the computation of the bit error rate (BER) curve and the peak signal-to-noise ratio (PSNR) of the reconstructed signal.

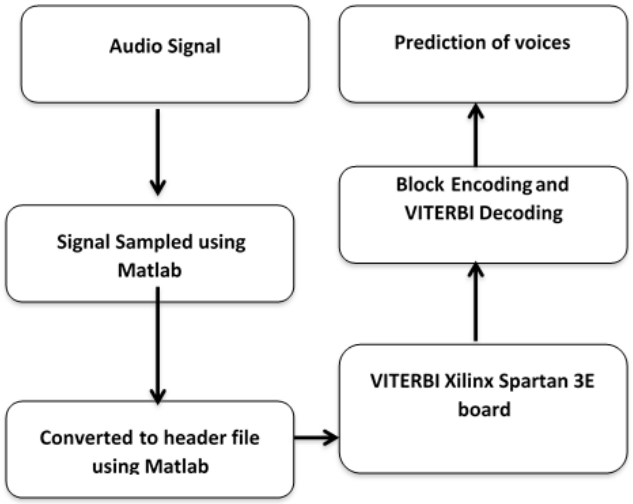

Figure 2. Overall implementation flowchart

3. RESULTS AND DISCUSSION

From the capture window we can observe that as voice signal of frequency 24 kHz is been multiplexed with PN sequence of sampled rate with 48 kHz to generate spreaded sequence as it will be suitable format to transmit over the network, the above two voice signals with different PN sequences to have two spreaded sequences with distinguish in manner. The first window Figure 3 shows the two voice signals with different consecutive levels of frequencies around 24 kHz to be a spreaded sequences and it's beside the waveform shows that spreaded sequences with different pn sequences as shown in the waveforms.

The QPSK modulation and demodulation of the spreaded signal as shown in Figure 4 with different waveforms, the spreaded signal will be again feeding to the processor to obtain the encoded output as convolution technique later it follows the signal was given to MATLAB for applying the QPSK modulation and the addition of AWGN noise through MATLAB.

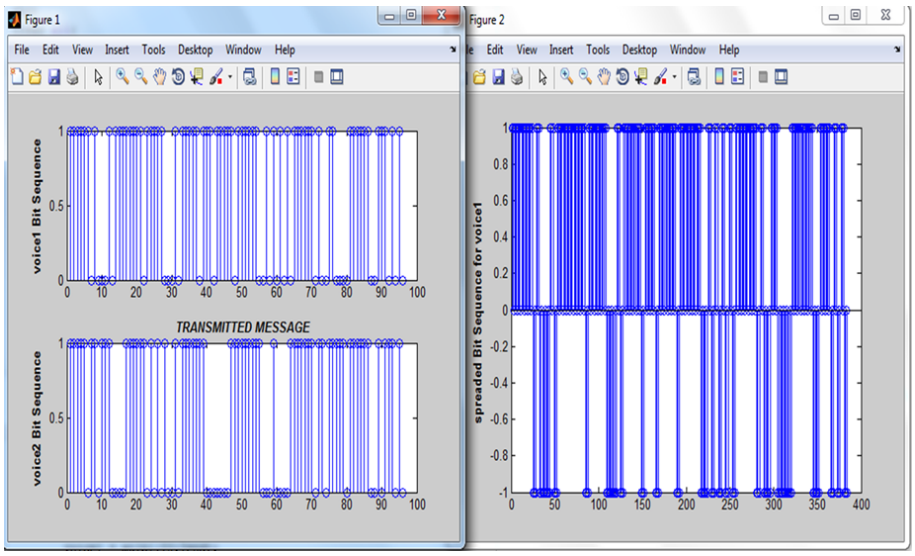

Figure 3. Output wave form results for two input signals voice1 and voice 2

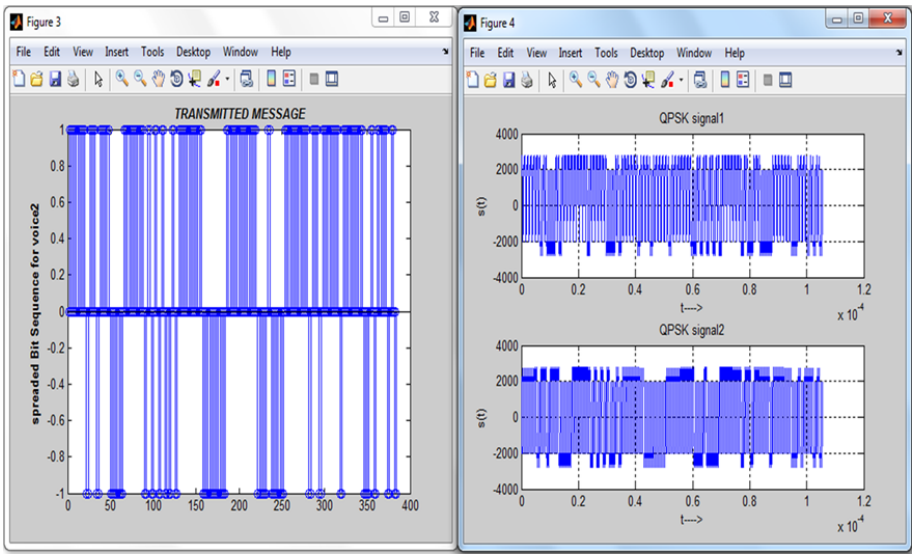

Figure 4. Output wave form results for spreaded sequences for voice 1 and voice 2

The Figure 5 shows the waveform of the QPSK demodulated waveforms that include the noise addition by manually with known amount of noise in the form the equated values in the MATLAB, particularly in QPSK demodulation the signal is mixed with the orthogonal basis functions in matched filters. The result obtained from matched filters are again tested with the decision devices, decision. Finally the outputs from decision devices are again multiplexed to get spreader message signal which undergoes de-spreading in CDMA receiver. Spread-spectrum in telecommunications usually a technique called signal structuring technique that makes direct sequence and frequency hopping on these signals. Similarly as Figure 6 shows that, the two de spreaded sequences of voice signal 1 and voice signal 2 can be used for multiple access or multiple functions, from the above received signal from the CDMA the de spreaded sequences for the voice signal 1 and voice signal 2 as shown below.

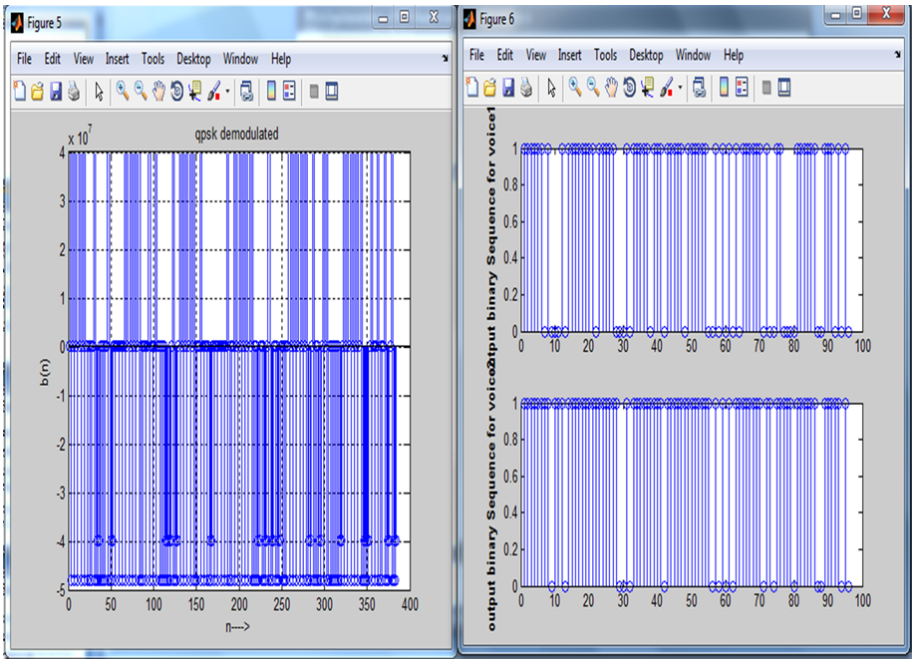

Figure 5. Output wave form results in binary sequence

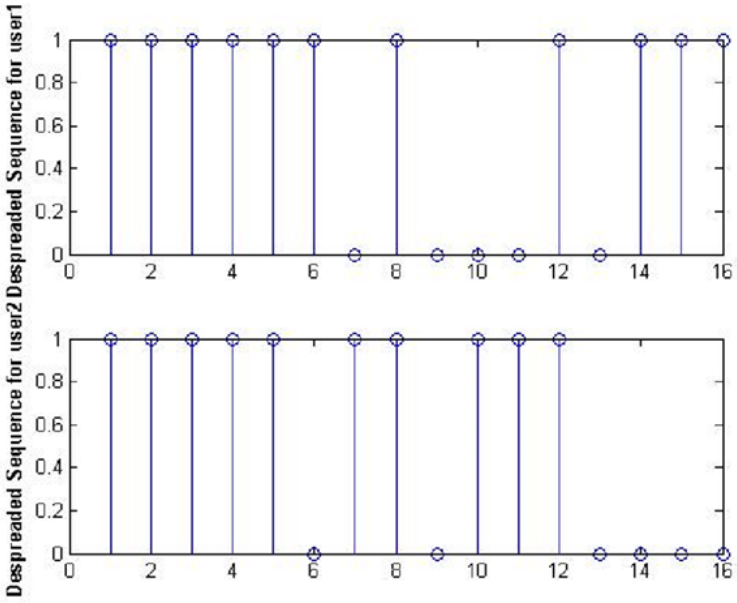

Figure 6. De-spreaded wave form signals in binary sequence

The demodulated signal is subsequently processed by the Viterbi algorithm for decoding, ensuring that the computational complexity does not exceed that of the conventional Viterbi decoder while progressively minimizing bit-level errors across iterations. In traditional wireless communication systems, transmitted signals occupy a fixed carrier frequency—typically in the megahertz or gigahertz range—which remains constant over time. For instance, when tuning to an FM radio station, the receiver displays the same carrier frequency at any instant. Because the carrier frequency in such systems is stable, the corresponding bandwidth remains confined within predefined limits, enabling straightforward signal identification and extraction by any intended receiver. In a similar manner, Figure 7 shows the waveform that illustrates the spreaded sequences of the first input voice signal and Figure 8 illustrates the spreaded sequence of the Second input voice signal transmitted within a specified frequency range, which defines the structure of the spread-spectrum sequence used in this implementation.

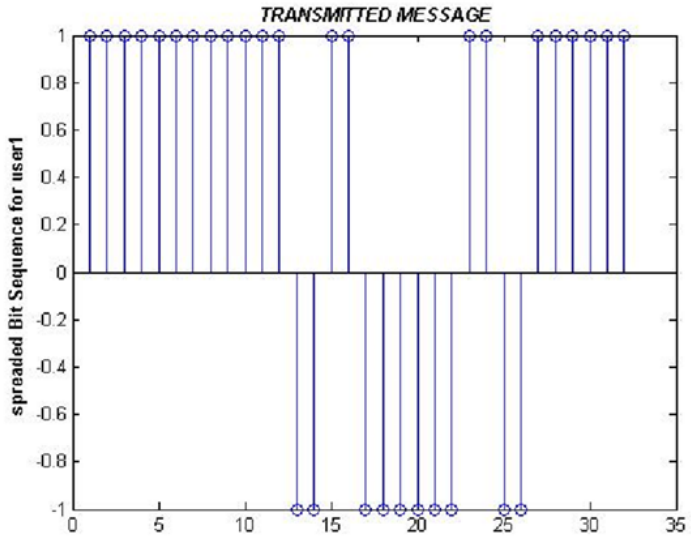

Figure 7. Spreadeed sequence wave form of binary sequence user1

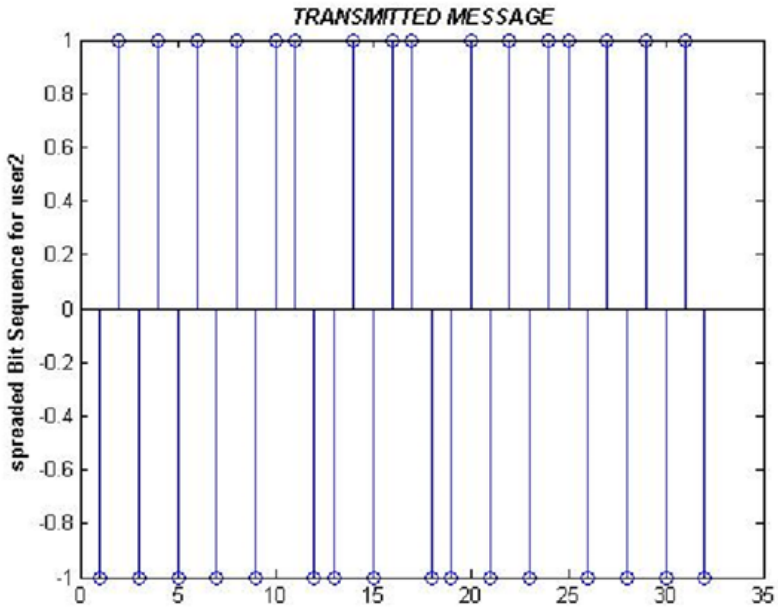

Figure 8. Spreaded sequence wave form of user2 signal

In Figure 9, QPSK modulation, two orthogonal sinusoidal carriers are employed to represent the modulation basis functions. The modulation process is carried out by altering the phase of these carriers in accordance with the incoming message symbols.

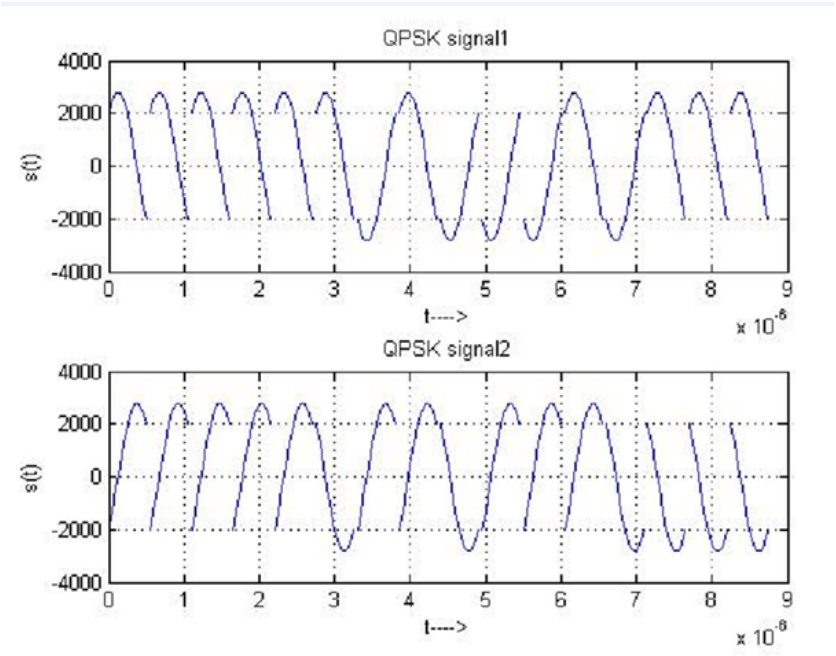

Figure 9. QPSK modulation of signal1 and signal2

In the QPSK demodulation stage, coherent detection is employed to ensure accurate recovery of the transmitted signal. This technique requires knowledge of the carrier frequency and phase at the receiver, which is achieved through the use of a phase-locked loop (PLL). The PLL synchronizes with the incoming carrier, maintaining lock on the frequency and tracking any variations in both frequency and phase. Once synchro-

nization is established, the received waveform is multiplied by the locally generated reference carriers for demodulation. The resulting demodulated signals, plotted at different operating frequencies, are illustrated in the Figure 10.

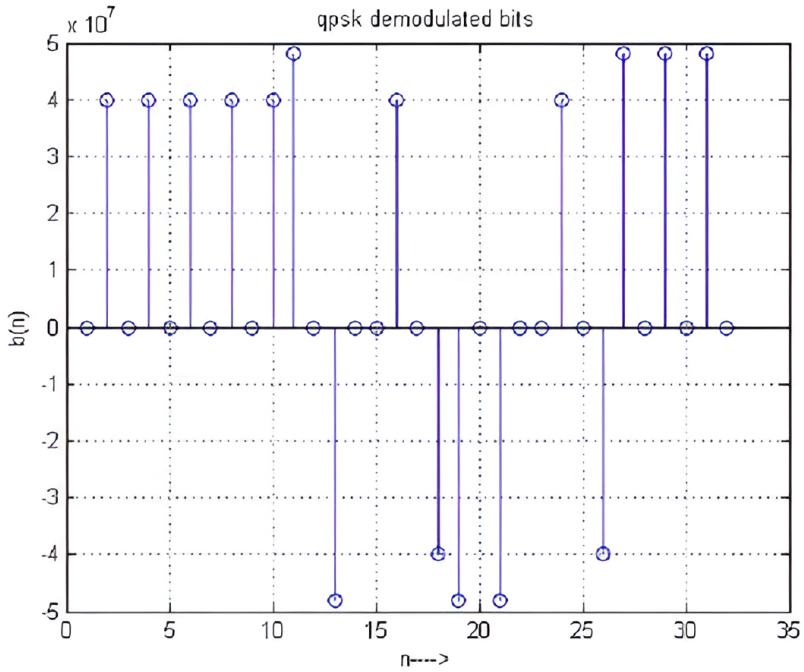

Figure 10. QPSK demodulation of signal1 and signal2

A MATLAB-based simulation was performed for the CDMA system in which the input signal was spread using a PN sequence. The spread signal was then transferred to the MicroBlaze processor to generate the convolutionally encoded data. This encoded output was subsequently returned to MATLAB, where QPSK modulation and AWGN channel noise were applied. The QPSK-demodulated signal was then fed back to the MicroBlaze processor for Viterbi decoding. The decoded data were passed through the CDMA de-interleaving stage, after which BER performance analysis was conducted. The corresponding results are summarized in Tables 1 and 2.

Table 1. Bit error reduction of voice signal for addition of 10 dB noise level with different code rates

Code rate	Voice input signal	Reduced error bit level by decoder	BER in percent (%)	BER in efficiency
$\frac{1}{2}$	6528	2024	33.76	66.33
$\frac{1}{3}$	6528	1332	20.40	79.66
$\frac{2}{3}$	6528	1908	29.22	70.81

Table 2. Tabulated results for Viterbi decoding of $\frac{1}{3}$ code rate for different noise level

AWGN in dB	Input signal bits	Error bits	BER in percent (%)	BER in efficiency
-10	179274	76321	42.57	66.33
-5	179274	46943	26.18	73.82
-3	179274	25864	0.144	99.855
5	179274	2	0.00001	99.9999

Figure 11 shows the plot of BER with signal to noise ratio in DB for ½ code rate results in MATLAB, from the graph we can say from this rate the BER achievement level is around 33%.

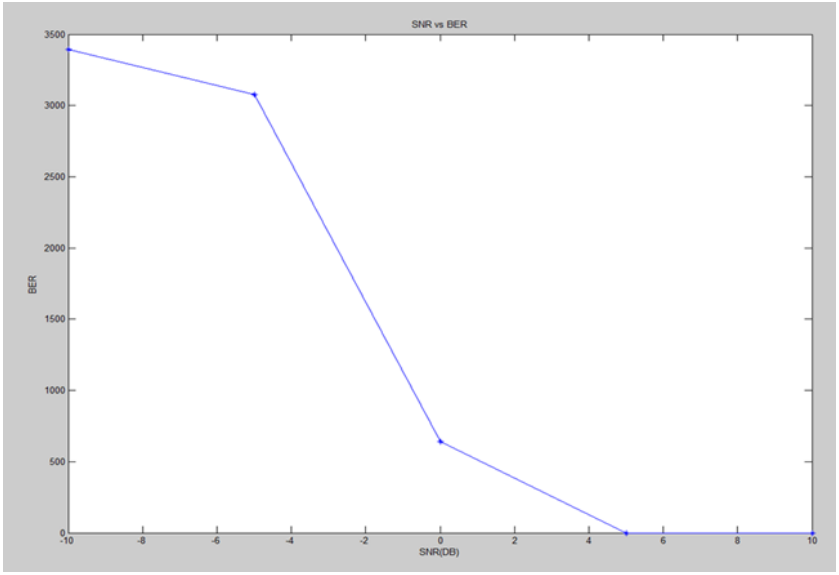

Figure 11. BER for $\frac{1}{2}$ code rate

Similarly the Figure 12 also shows the plot of BER with signal to noise ratio in dB for 1/3 code rate results drawn in MATLAB, from the graph we can say from this code rate the BER achievement level is around 29%, hence in comparing to ½ code rate reduction of error rate is higher.

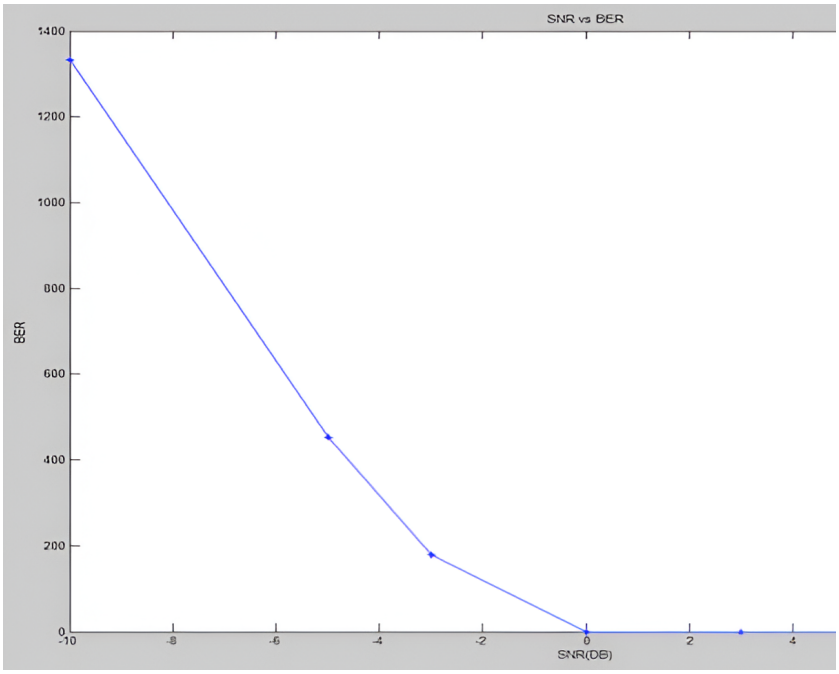

Figure 12. BER for $\frac{1}{3}$ code rate

Similarly the Figure 13 snap also shows the plot of BER with signal to noise ratio in dB for 2/3 code rate results drawn in MATLAB, from the graph we can say from this code rate the BER achievement level is around 29%, so we can conclude from the above observation among all three code rates the ½ code rate is the best as it is higher rate in reduction of error bits.

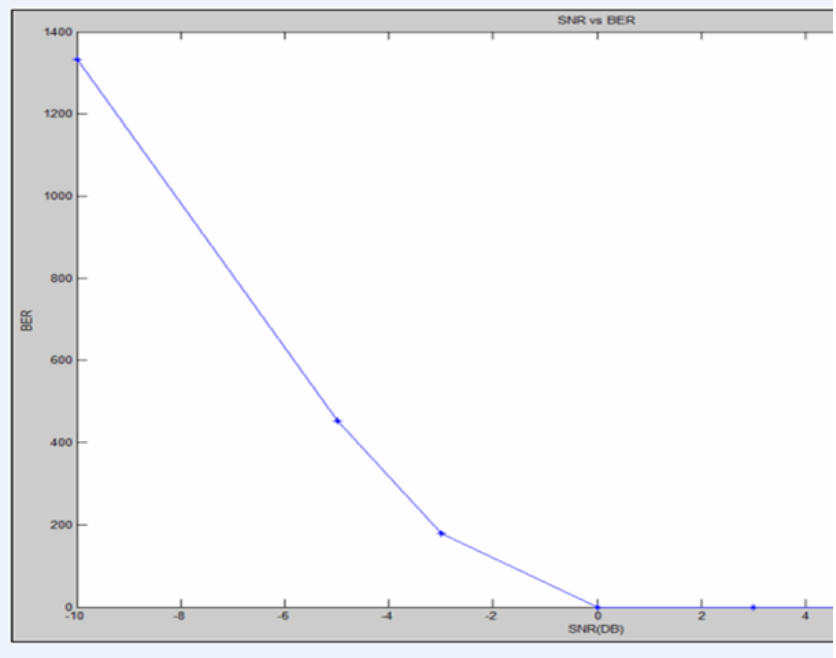

Figure 13. BER for $\frac{2}{3}$ code rate

4. CONCLUSION

Based on the above literature survey, analysis and development for implementing the above mentioned concept and implementation of the multiuser CDMA in MATLAB and the encoder and decoder implementation on Micro blaze processor followed with the QPSK modulation and demodulation on MATLAB, we can conclude from the results obtained as tabulated in section 5 for the encoder followed by Viterbi decoder with Microblaze processor, it is concluded that even for a very long data of length about more than one lakh samples taken from the voice signal and analyzed in terms of how BER varies with change in AWGN noise level for ½, 1/3 and 2/3 code rates. This paper has also analyzed the error reduction with the cumulated order on sample selection. This infers that the BER has reduced very much 99.99% while using ½ encoding code rate.

FUNDING INFORMATION

Authors state no funding involved.

AUTHOR CONTRIBUTIONS STATEMENT

This journal uses the Contributor Roles Taxonomy (CRediT) to recognize individual author contributions, reduce authorship disputes, and facilitate collaboration.

Name of Author	C	M	So	Va	Fo	I	R	D	O	E	Vi	Su	P	Fu
Ramjan Khatik	✓	✓	✓	✓	✓	✓		✓	✓	✓			✓	
Afzal Shaikh		✓				✓		✓	✓	✓	✓	✓		
Shraddha Sawant		✓				✓		✓	✓	✓	✓	✓		
Pritika Patil		✓				✓		✓	✓	✓	✓	✓		

C : **C**onceptualization
M : **M**ethodology
So : **So**ftware
Va : **Va**lidation
Fo : **Fo**rmal Analysis
I : **I**nvestigation
R : **R**esources
D : **D**ata Curation
O : Writing - **O**riginal Draft
E : Writing - Review & **E**diting
Vi : **Vi**sualization
Su : **Su**pervision
P : **P**roject Administration
Fu : **Fu**nding Acquisition

CONFLICT OF INTEREST STATEMENT

The authors declare that the research was conducted in the absence of any commercial or financial relationships that could be construed as a potential conflict of interest.

DATA AVAILABILITY

The data supporting the findings of this study are available upon reasonable request from the corresponding author. Due to the nature of the research, some of the data may be subject to confidentiality agreements and may not be made publicly available.

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BIOGRAPHIES OF AUTHORS

Ramjan Khatik received his Bachelor degree in Electronics Engineering. He received the M.E. degree from Ramrao Adik Institute of Technology, University of Mumbai in 2011, and the Ph.D. degree from Visvesaraya Technological University (VTU), in 2023. He is currently a Director and Professor of Electrical and Computer Engineering with the Anjuman-I-Islam's Kalsekar Technical Campus, University of Mumbai, New Panvel, Navi Mumbai, India. He can be contacted at email: rak.ktc@gmail.com.

Afzal Shaikh graduated from University of Mumbai, India, in 2002. He received the M.E. degree from Ramrao Adik Institute of Technology,University of Mumbai in 2011, and the Ph.D. degree from Visvesaraya Technological University (VTU), in 2022. He is currently a Associate Professor of Electrical and Computer Engineering with the School of Engineering and Technology, University of Mumbai, New Panvel, Navi Mumbai, India. He can be contacted at email: afzal.aiktc@gmail.com.

Shraddha Sawant completed her B.E. and M.E degrees in electrical engineering from the University of Mumbai, Mumbai, India, in 2011 and 2014, respectively. Since 2020, she has been a research scholar in the Department of Electrical Engineering, Fr. C. Rodrigues Institute of Technology, Navi Mumbai, Mumbai, India. Since 2014, she has been a member of the Faculty of Department of Electrical and Computer Engineering with the School of Engineering and Technology University of Mumbai, New Panvel, Navi Mumbai, India. She can be contacted at the email: shraddha.hule@aiktc.ac.in.

Pritika Patil graduated from Savitribai Phule, Pune University, India in 2012. She received the ME degree from Savitribai Phule, Pune University, India in 2017 and pursuing Ph.D degree from Ramrao Adik Institute of Technology. She is currently as a Assistant Professor of Electrical and Computer Engineering with the school of Engineering and Technology University of Mumbai, New Panvel, Navi Mumbai, India. She can be contacted at the email: pritikapatil@aiktc.ac.in.