Analysis of channels for Software Defined Radio using LabVIEW

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***Abstract*— The role of radio in wireless communication has been extended to a much extent from few decades. The basic concept of software defined radio is that the radio can be configured from software hence reducing the need of hardware. In this paper software defined radio (SDR) 32-QAM modem is implemented using LabVIEW software. LabVIEW is a graphically based programming language developed by National Instruments. It has built in functionality for test and measurement automation, instrument control, data acquisition, and data analysis applications. The aim of this paper is to simulate SDR for next generation wireless communication having Gaussian and time varying channel. The simulation is done to analyze the performance of hamming coding in Gaussian channel and convolution coding in time varying channel is compared Adaptive filter is used to minimize the effect of noise. It is obvious from the simulation results that the SNR and BER values are analyzed using hamming coding & convolution coding depending upon applications & thus gives better results**.

***Keywords*— Software Defined Radio, LabVIEW, QAM Modem, Gaussian Channel and Adaptive Filter.**

I. Introduction

This paper discusses Software Defined Radio 32-QAM Modem for Gaussian Channel and time varying channel, using hamming coding in Gaussian channel and convolution coding in time varying channel is implemented using LabVIEW. Software-Defined Radio is a new technology in the wireless space that will enable its users to receive and transmit services and features upgrades almost instantly. The basic concept of the SDR software radio is that the radio can be totally configured or defined by the software so that a common platform can be used across a number of areas and the software used to change the configuration of the radio for the function required at a given time, thus eliminating the complex and expensive hardware upgrades. This enables the user to talk and listen to multiple channels at the same. The role of modulation techniques in software radio is thus very important as the modulation techniques forms the core part for any wireless system. The SDR system consists of an RF section, IF section, and baseband section. RF section consists of analog modules connected to ADC and DAC both of which form the IF section for receive and transmit functionality. The third section is the baseband section which performs baseband operations, connection setup, equalization, timing recovery, correlation, error correction coding, etc. The front end in an SDR system consists of all components used to transmit and receive, process, and down convert RF.[1][2] The back end or baseband section is the signal processing functionality consisting of general purpose processors. These processors run the software that implements various functions of the communication system, including GSM or CDMA.

In this paper, 32-QAM (Quadrature Amplitude Modulation) is chosen to be the modulation scheme of the designed software defined radio system as it is widely used for data transmission applications such as high speed cable, FAX modem, multi-tone wireless, microwave digital radio, DVB-C (Digital Video Broadcasting Cable), Modems and Satellite channels etc. QAM takes advantage of the fact that it is possible to send more number of bits as compared to other modulation process such as ASK, PAK, etc. and also two different signals simultaneously on same carrier frequency. Different coding techniques are used depending upon the applications [3]. In this paper hamming coding is used in Gaussian channel and convolution coding is used in time varying channel and the performance is analyzed. The unwanted noise is removed using adaptive filtering.

II. PREVIOUS WORK

Software defined radios have their origins in defense sector since late 1970's in U.S and Europe. During that phase VLF radios which were based on ADC and connected to an 8085 microprocessor were used by the ground forces of USA and Great Britain. The history of SDR began in the mid 1980's. The term Software radio was coined by Joseph Mitola in 1991 to refer to the class of reprogrammable or reconfigurable radios whose first paper (SpeakEasy) was published in 1992. In 1984 E-Systems coins "software radio" term: E-Systems, now Raytheon, coined the term "software radio" in a company newsletter. It referred to a prototype digital baseband receiver equipped with an array of processors that performed adaptive filtering for interference cancelation and demodulation of broadband signal. In 1991 SPEAK easy: The first military program that specifically required a radio to have its physical layer components implemented in software was DARPA's SPEAKeasy. Its primary objective, originating from the U.S. Air Force, was to have a single radio that could support ten different military radio protocols and operate anywhere between 2 MHz and 2 GHz. A secondary goal was the ability to incorporate new protocols and modulations, thereby future-proofing the radio hardware.

In 1997 Creation of the JTRS: The Joint Tactical Radio System was created by the U.S. Department of Defense to increase interoperability. 2001 GNU Radio: Evolved from an MIT-originating framework called PSpectra, GNU Radio is an open-source framework for the development of SDR applications within a PC environment. With more than 5,000 claimed users as of 2012, it is by far the most popular SDR development toolset. Complete waveforms such as P25, 802.11, ZigBee, Bluetooth, RFID, DECT, GSM, and even LTE (still a work in progress) can be downloaded from the repository and run on any x86 system.

In 2009 paper "Modulation Technique for Software Defined Radio Application" was published by Muhammad Islam, M. A. Hannan, S. A. Samad & A. Hussain. which discussed PSK as modulation technique for Software Defined Radio and evaluated BER for Software Defined Radio. Another paper was published for SDR using QAM Modulation Technique for RFID which concludes we can use the QAM as modulation scheme for better transmission performance.

The multi-billion dollar program was very ambitious and experienced difficulties, delays, and cost overruns. It was officially cancelled in 2011 by the U.S Under Secretary of Defense, who stated that the products and technologies resulting from the program were unlikely to meet the established requirements. A paper published by N. KEHTARNAVAZ for Software Defined Radio implemented using LabVIEW considered QAM as its modulation scheme, also considered the channel to be ideal. But channel can't be ideal practically. So in this paper simulation for SDR is done for Gaussian channel using hamming coding technique and time varying channel using convolution coding technique. The results are analyzed and the effect of noise is minimized using adaptive filtering technique.

III. SOFTWARE DEFINED RADIO 32-QAM MODEM USING HAMMING CODING FOR GAUSSIAN CHANNEL

The building blocks of the 32-QAM modem system in LabVIEW are as shown in Figure.1. This system consists of the Transmitter and the Receiver. The Transmitter is formed of four modules i.e. Message source, Pulse Shape Filter, Hamming /Convolution Encoder, 32-QAM Modulator and Receiver is formed of rest of modules i.e. Hilbert Transformer, 32-QAM Demodulator, Hamming/Convolution Decoder, Sync & Tracking [4][5].

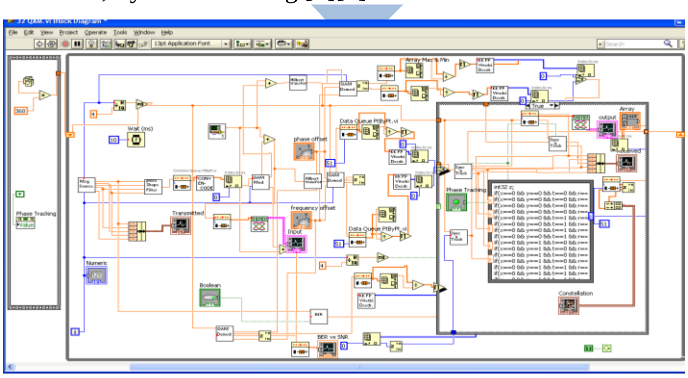

Figure 1: Block Diagram of 32-bit QAM Model system

A. Message Source

The first component of QAM Modem is the Message Source. In this the MLS generates a maximum length sequence of ones and zeros [6].

The frame marker is a distinct pattern of bits that never occurs in the stream for message data.

known bit sequence of length 10 is used as the frame marker in this VI block [4][7].

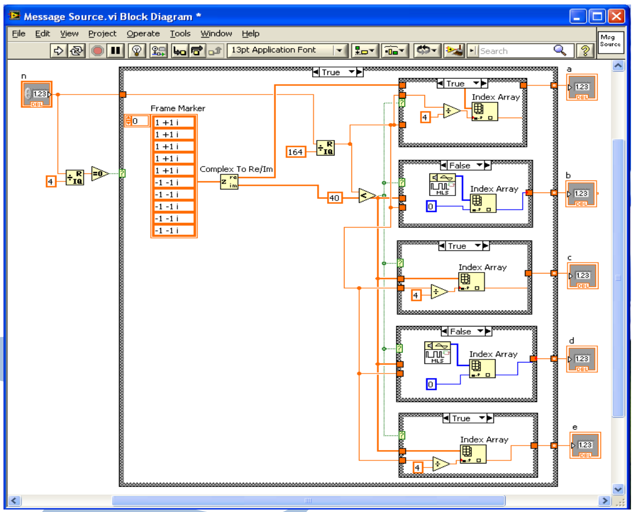

Figure 2: **Message Source**

The frame maker is so chosen to carry out low correlation with the PN sequence. The generated samples are oversampled 4 times according to the specification of the pulse shape filter. It is done by comparing with 0 the remainder of a global counter. For the Remaining three executions of this VI zero samples get generated the type of pseudorandom binary sequence. 10 complex numbers is used as a constant array to specify the marker bits. The real parts of the complex values are used as the frame marker bits of the in-phase components while the imaginary components is used as the frame marker bits of the Quadrature-phase samples[9]. The Message Source VI is shown in Figure 2

Pulse Shape Filter

The generated message sequences are passed through a raised cosine FIR filter to create a band-limited baseband signal. The excess bandwidth beyond the Nyquist frequency is specified by a roll-off factor of the filter. A roll-off factor of 0.5 is used in our implementation. It is done before the process of modulation by changing the waveform of the transmitted pulse. It is done to make the transmitted signal better suited for the communication channel. This transmitting signal with high modulation rate when made to pass through a band limited channel can create ISI. By using pulse shape filtering ISI/ noise caused by the channel can be controlled [7][9]. The message sequences generated is passed through a raised cosine FIR filter to create a band-limited baseband signal. Hence a raised cosine filter is used as a pulse shape filter. An FIR filter is used for this purpose. The Pulse Shape Filter VI is shown in Figure 2. Filter to create a band-limited baseband signal. Hence a raised cosine filter is used as a pulse shape filter. An FIR filter is used for this purpose. The Pulse Shape Filter VI is shown in Figure 2.

A. QAM Modulator

QAM can be considered as the logical extension of QPSK. The two independent signals are transmitted over the same medium at the same time [7]. QAM modulation process involves the multiplication of complex carriers having same frequency but differ in phase by 90^0.The QAM Modulator VI is shown in figure 3.

B. ADAPTIVE Filter

Adaptive Filter is used in many applications, such as noise cancellation. The coefficients of an FIR filter are adjusted using filter coefficients according to an error signal in order to adapt to a desired signal. The VI for Adaptive Filter is as shown in figure 4.

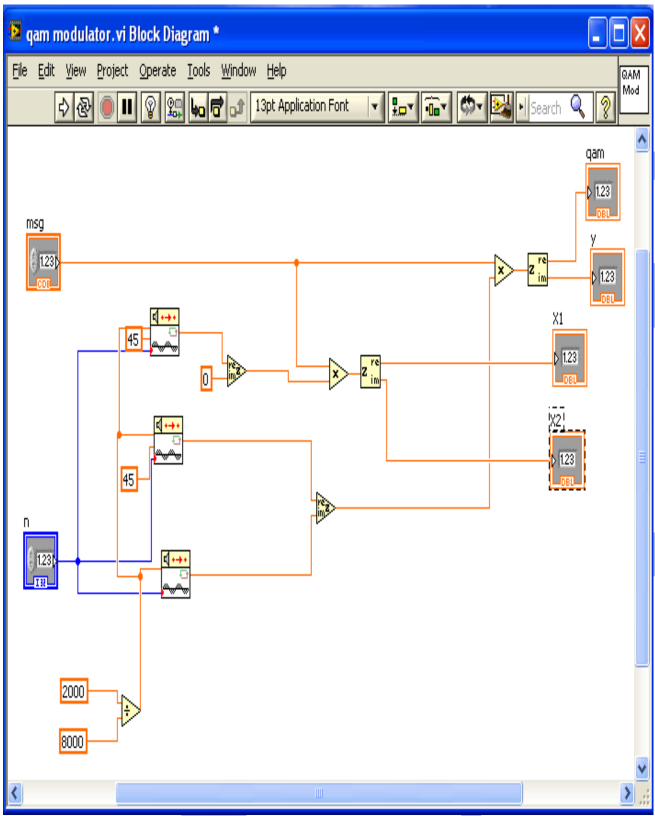

Figure 3: **32-QAM Modulator**

An adaptive filter is a filter that self-adjusts its transfer function according to LMS (Least Mean Square) algorithm [4]. As the optimizing algorithms are complex, most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal. A feedback connection is provided to transfer data from one iteration to next inside a For Loop or a While Loop. This module introduces the process of adaptive filters by using the LMS algorithm. The adaptive filter adjusts its coefficients to minimize the mean-square error between its output and that of an unknown system. Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal [8] [9].

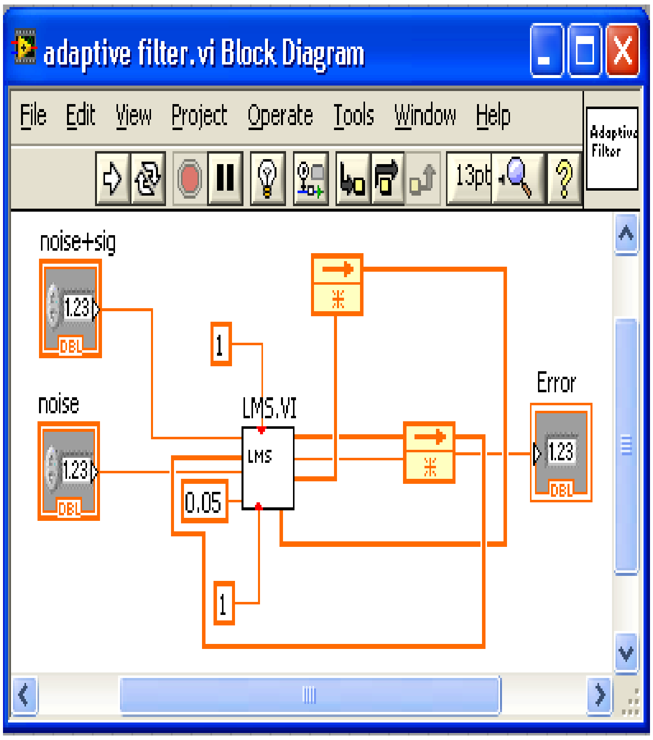

Figure 4: **Adaptive Filter VI.**

A. Hilbert Transformer

A Hilbert transformer is used to build the analytic signal for demodulation from the transmitted QAM signal. That is, if $r(nT)$ is considered to be the sampled received signal, the analytic signal $r(nT) +$ is given by

$r(nT)\ r(nT)\ j\hat{r}(nT) + = +$ (4)

where $\hat{r}(.)$ indicates the Hilbert transform of $r(.)$ An FIR filter is used for its implementation [11]. The VI for Hilbert Transform is shown in figure 6.

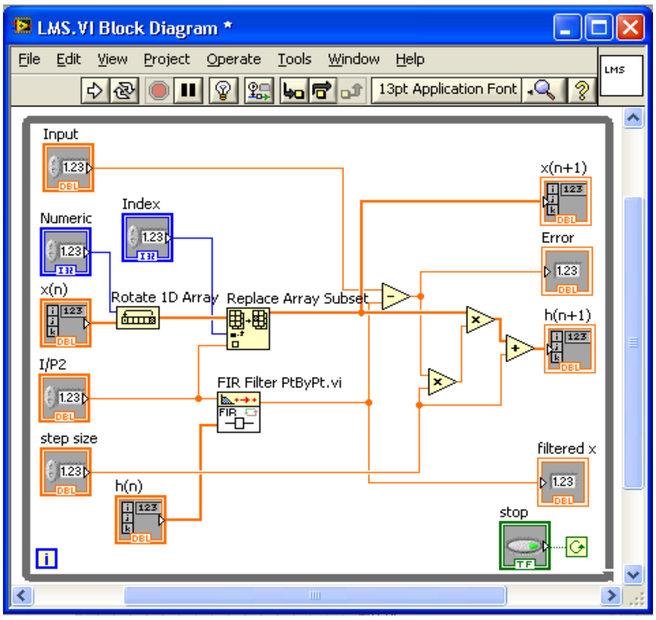

Figure 5: **LMS VI**

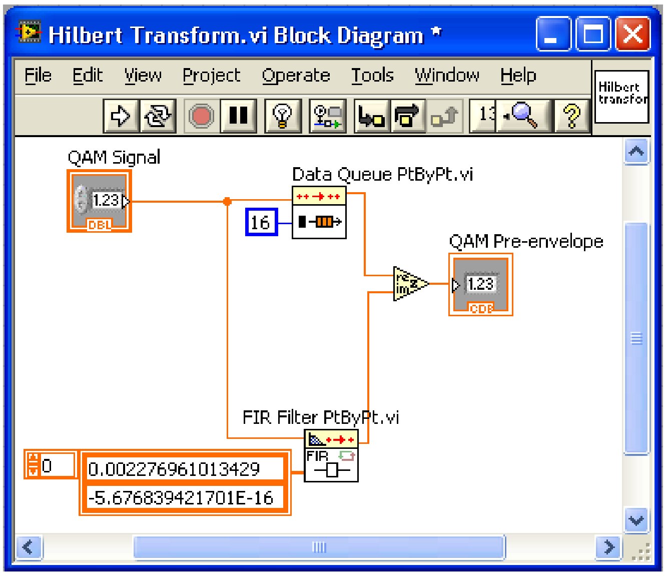

Figure 6: **Hilbert Transform VI.**

B. QAM Demodulator

This block involves the multiplication of a complex carrier having a negative frequency with the analytic signal obtained from the Hilbert transformer block. The Hilbert transformer is thus implemented here as a band pass filter. An even number, 32, is specified as the filter order, to have an integer group delay [10][7].

The complex envelope of the received QAM signal can be expressed as

$$\tilde{r}(nT) = r_{+}(nT)e^{-jw_c nT}$$

$$= a(nT) + jb(nT)$$

The Hilbert transformer gives the analytical signal which is demodulated by the QAM demodulator as shown in Figure 7.

C. Sync & Tracking VI – Frame Synchronization Mode

Sync & Tracking VI is used for Frame synchronization and Phase/Frequency tracking

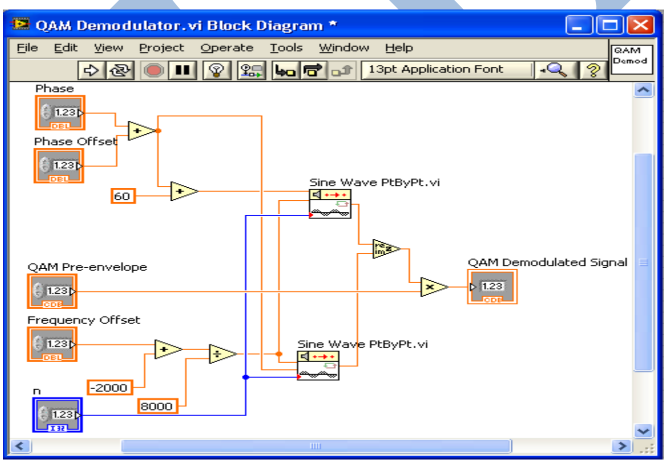

Figure 7: **QAM Demodulator**

Synchronization is an act of synchronizing that means concurrence of events with respect to time. In block diagram shown in figure 8, the input samples are passed through Complex Queue point by point VI, which creates a data queue of complex numbers to obtain beginning of frame [8][11]. A case structure will not be executed until the queue is completely filled. Extra 16 bits are added to add delays related to the filtering operations in the transmitter. A counter is used to count number of samples filling up the queue. A Boolean is a primitive data type that can have any one of the two values: TRUE or FALSE. The VI for sync and tracking is as shown in figure 8

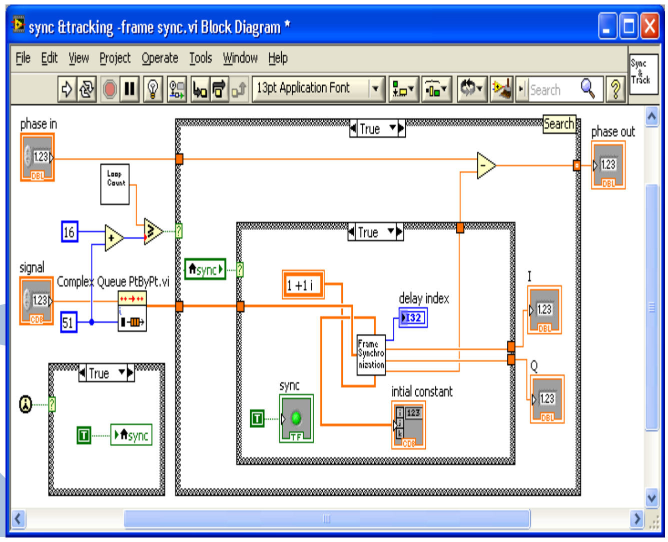

Figure 8: **Sync & Tracking VI – Frame synchronization Mode**

A. Sync & Tracking VI – Phase and Frequency Tracking

Frame synchronization is required for properly grouping the transmitted bits into an alphabet. To achieve this synchronization, a similarity measure, consisting of cross correlation, is calculated between the known marker bits and the received samples. Now after getting the decision based carrier tracking is used to examine the phase offset between transmit and received signal [11][14]. Next LMS update method is used to minimize the phase error in the signal. When both phase and frequency tracking are considered, the carrier phase of the receiver becomes the phase update. The sync and tracking VI- phase and frequency tracking mode VI is shown in figure 9

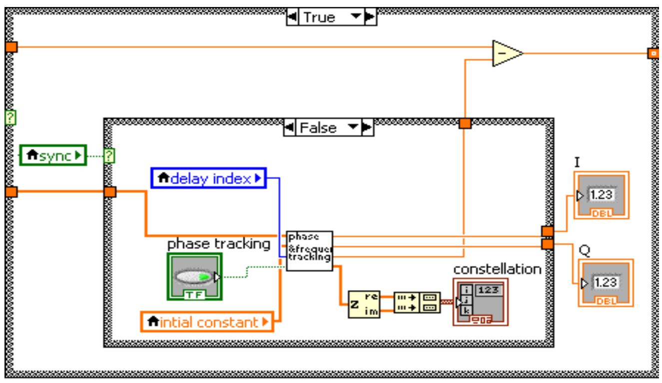

Figure 9: **Sync & Tracking VI – Phase and Frequency Tracking Mode VI**

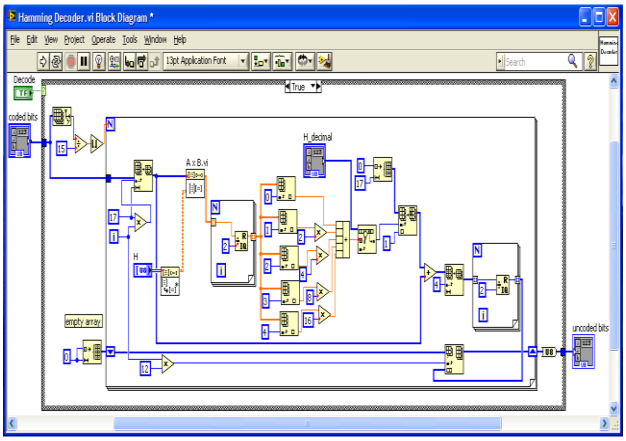

Figure 10: **Hamming decoder**

E. Hamming Decoding

The Hamming Decoder can correct maximum 1 bit every 17 bits of codeword input which is a function of the minimum Hamming distance between the possible code words. The Decoding can be expressed as Syndrome Decoding. Decoding is computed when a symptom matrix is computed which is the received bit sequence multiplied by a parity check matrix[12].

The parity check matrix has the special property that the multiplication of any codeword with the parity matrix results in a zero vector [13]. The Hamming Decoder VI is shown in

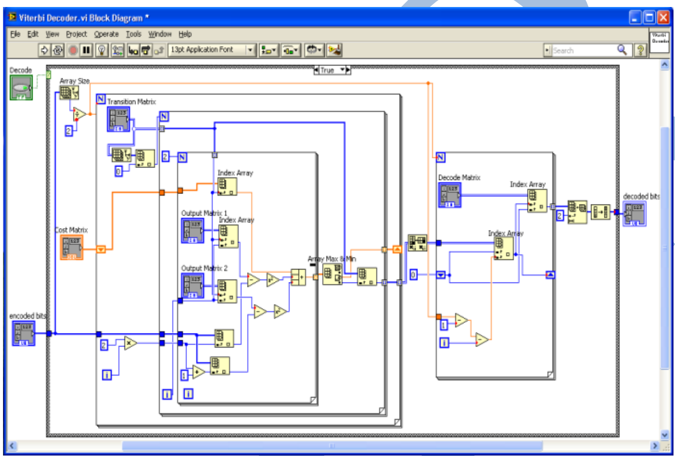

Figure 10.

F. Viterbi Decoding

Viterbi decoding is an optimal (in a maximum-likelihood sense) algorithm for decoding of a Convolution code as this simplifies the decoding operation. The decoder is a Viterbi decoder which then solves for the globally optimum bit sequence. The algorithm updates a path cost as it steps through each stage of the possible output sequences. At each state, it also calculates the likelihood of entering each possible new state based on the cost of the previous state. The algorithm then needs two additional zero bits after every sequence in order to force the encoder back into the zero state and to assume that the encoder ends at the all zero state [13][14]. These two tail bits represent a fractional loss rate between the coded and that of encoded bit sequence. The Viterbi Decoder VI is shown in

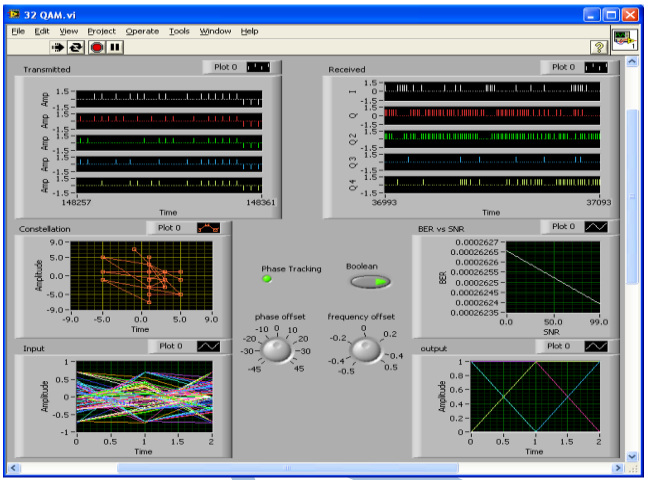

Figure 11. **Viterbi decoder**

IV. SIMULATION RESULTS

This section provides the simulation study done to test the performance of SDR with hamming coding using Gaussian channel and convolution coding using time varying channel. The simulation results can be observed from a waveform chart and an XY graph from modem systems shown in figure 12. The amplitudes of some of the received samples become too small as the constellation of the received signal is rotated. The simulation is done to analyze the performance of Gaussian channel using coding and time varying channel using convolution coding. Simulation results can be observed from waveform charts and an XY graph for 32-QAM Modem system as shown in figure 12. beneficial compared to that of PSK techniques in terms of increased data rate.

Figure 12**: Received Signal for SDR using Convolution Coding**

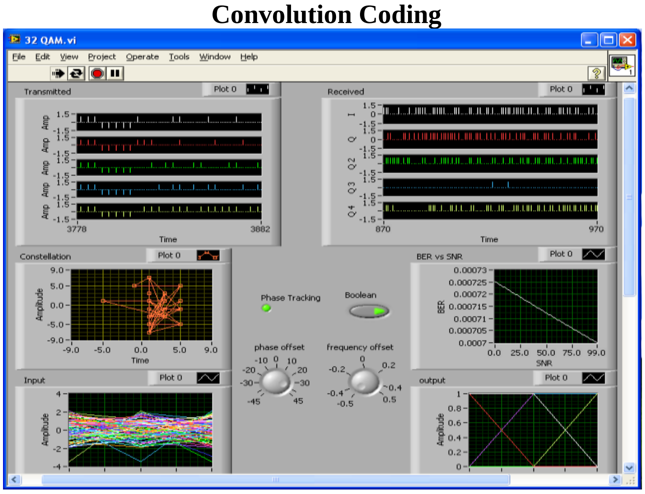

Figure 13**: Received Signal for SDR using Hamming Coding**

Hence the received signal becomes a perfect reproduction of the transmitted signal by correcting the erroneous bits present in channel using encoding and decoding techniques. Thus the BER achieved using Convolution Coding is much less compared to that achieved with Hamming Coding algorithm at a particular value of SNR for the designed SDR Modem system.

V. CONCLUSION

In this paper, 32-bit QAM modem system for Gaussian channel using hamming coding and time varying channel using viterbi coding consisting of a Message Source, a Pulse Shape Filter, QAM Modulator, Adaptive Filter, a Hilbert Transformer, QAM Demodulator, Sync & Tracking for frame synchronization and phase/frequency tracking are built in graphical programming software LabVIEW allowing software-defined radio system to be built in much less time. The result of simulation is optimum using adaptive equalizer to remove external interference and comparable to the result given by LabVIEW-Based Software-Defined Radio. The two coding techniques in different type of channels are implemented and the results are compared. Hence, the designed Software Defined Radio (SDR) 32-bit QAM Modem can be practically implemented

VI. REFERENCES

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