

CONCATENATED CODING IN OFDM FOR WIMAX USING USRP N210 AND GNU RADIO

B. Siva Kumar Reddy¹ and B. Lakshmi²

Research Scholar¹, Associate Professor²
Department of Electronics and Communication Engineering
National Institute of Technology Warangal, Andhra Pradesh-506004, India.

ABSTRACT

A software Defined Radio (SDR) device employs a reconfigurable hardware (Universal Software Radio Peripheral-USRP) that may be programmed over-the-air or software (GNU Radio) to function under different Wireless standards. This paper analyzes the effect of various parameters such as channel noise, frequency offset, timing offset, timing beta, FLL (Frequency Lock Loop) bandwidth, Costas loop (phase) bandwidth, filter roll off factor and multiply const on OFDM signal in WiMAX physical layer with concatenated coding using SDR test bed. Concatenated coding is performed by suggesting RM coder and Convolutional coders as inner code and outer codes respectively. Moreover, bit error rate and symbol error rates performance are analyzed by varying bits per symbol, window size and modulation scheme. Results proved that BER and SER values are improved as modulation scheme size (M) is increased. OFDM signal transmission and reception is performed using USRP N210 and configured by GNU radio in the laboratory environment.

KEYWORDS

BER, GNU Radio, OFDM, SDR, USRP, WiMAX.

1. INTRODUCTION

Software Defined Radio (SDR) [1] is a term reinvented from software radio by Joseph Mitola in 1991, while recognizing the possibilities of re-configurability and re-programmability of radio systems. The idea behind software defined radio is to perform all signal processing functions with software instead of using dedicated circuitry. The most obvious benefit is the reduction in complexity and cost because of less hardware usage. An ideal SDR would have all the radio-frequency bands and modes determined software-wise, meaning it would comprise only of an antenna, DAC or ADC and a programmable processor. However, in practical systems, the RF front-end has to be enforced as well in order to support the receive/transmit mode. In this paper SDR is implemented by employing USRP (Universal Software Radio Peripheral) N210 [2] as hardware and GNU radio [3] as software platforms.

WiMAX (Worldwide Interoperability for Microwave Access) [4] is one of the most widely using broadband wireless access technologies based on the IEEE802.16 standard for Metropolitan Area Networks (MAN). WiMAX supports fixed and mobility services called as Fixed WiMAX (IEEE 802.16d) [4] and Mobile WiMAX (IEEE 802.16e-2005) [4] respectively. For mobile communications below 6 GHz frequencies have good propagation properties and are better suitable. 802.16 allows for several antennas to be employed at the transmitter and the receiver to provide a MIMO [5] system.

The field of channel coding [6] is pertained with transmitting a stream of data at as high a rate as possible over a given communications channel, and then decoding the original data reliably at the receiver, employing encoding and decoding algorithms that are executable to carry out in a given technology. The motivation for concatenating two coding schemes is to achieve large coding gains with affordable decoding complexity. In coding theory, concatenated codes [6] form a class of error-correcting codes that are gained by combining inner and outer codes. In this paper concatenated coding structured as Convolutional coding [6] as outer code and Reed Muller coding [7] as inner code.

The rest of the paper is structured as follows: Section 2 demonstrates the experimental setup of SDR with USRP N210 and PC (GNU Radio Companion (GRC)) and gives the clear explanation about USRP and GNU Radio platforms with specifications. Section 3 presents the WiMAX physical layer with working principles and explains each block which are used in GRC and for detailed information can refer ref [3]. Section 4 delivers observed experimental results and corresponding figures. Section 5 concludes the paper from the results obtained from Section 4.

2. EXPERIMENTAL SETUP

SDR comprises of RF section, IF section and baseband processing section. RF and IF sections are incorporated in USRP and baseband processing is performed in a PC using GNU radio companion (shown in Figure. 1). The USRP N210 [2] allows for high-bandwidth, high-dynamic range processing capability. This includes a Xilinx® Spartan® 3A-DSP 3400 FPGA (Field Programmable Gate Array), two 100 MS/s ADCs, two 400 MS/s DACs and Gigabyte Ethernet connectivity to flow information to and from host processors. The USRP N210 adds a larger FPGA than the USRP N200 [2] for additional logic, memory and DSP resources based demanding applications. All baseband signal processing (e.g. modulation, amplification, mixing, filtering etc.) is done in GNU Radio [3]. USRP can be reconfigured (in runtime also) to desired specifications in host computer by using GNU Radio. GNU Radio is a free software development toolkit that offers the signal processing runtime and readily available more than 100 processing blocks to implement software radios employing low-cost external RF hardware (USRP) and allows real time SDR applications [1]. In GNU Radio, signal processing blocks are written in Python and those are connected using C++ and both languages are communicated by SWIG (Simplified Wrapper and Interface Generator) interface compiler. Thus, the developer is allowed to accomplish real-time, high-throughput radio systems in a simple to-use, rapid-application development environment. In this paper all GNU Schematics (Signal flow graphs) are drawn for Mobile WiMAX specifications (FFT size=1024) [4].

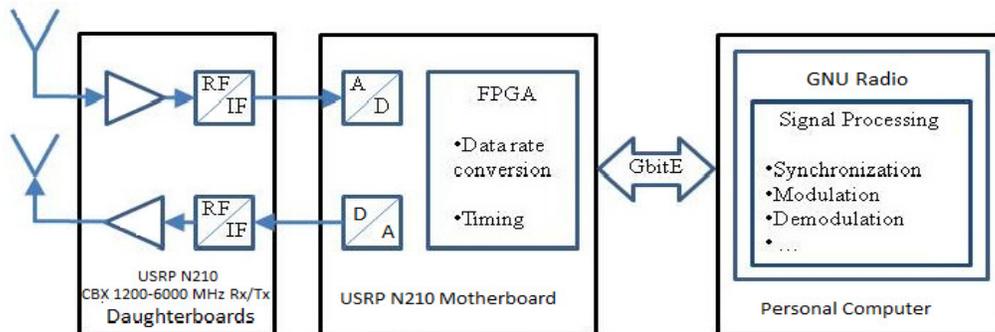


Figure 1: Software Defined Radio block diagram with USRP N210 and GNU Radio.

3. WIMAX PHYSICAL LAYER

The function of the PHY layer is to encode the binary digits that symbolize MAC frames into signals and to send and obtain these signals throughout the communication media. The WiMAX PHY layer is based on OFDM (Orthogonal Frequency Division Multiplexing)/OFDMA (Orthogonal Frequency Division Multiple Access) technologies [8] which are applied to enable high-speed data, video, and multimedia communications and is employed by a variety of commercial broad band systems. The WiMAX PHY layer (shown in Figure. 2) includes various functional stages: (i) Forward Error Correction (FEC): including; scrambling, concatenated encoding and interleaving (ii) OFDM modulation and (iii) Receiver synchronization.

The data flow processing through physical layer is described as follows. A signal with 6 GHz frequency is captured from the environment by using CBX daughterboard (in USRP N210) and GNU radio. The captured 6 GHz signal is passed to scrambler and it scrambles an input stream employing an LFSR (Linear Feedback Shift Register) [6]. This block influences on the LSB only of the input data stream, i.e., on an "unpacked binary" stream, and develops the same format on its output. The CCSDS encoder block [3] executes convolutional encoding [6] applying the CCSDS standard polynomial ("Voyager"). The input and output are an MSB first packed stream of bits and a stream of symbols 0 or 1 representing the encoded data respectively. Since the code rate is $1/2$, there will be 16 output symbols for every input byte. This block is planned for continuous data streaming, not packetized data. There is no provision to "flush" the encoder.

Data interleaving is used to increase efficiency of FEC by disseminating burst errors inserted by the transmission channel over a long time. The interleaving is determined by a two step permutation. First checks that adjacent coded bits are mapped onto nonadjacent subcarriers. The second permutation checks that adjacent coded bits are mapped alternately onto less or more significant bits of the constellation, thus eliminating long runs of lowly reliable bits. The first permutation is given by

$$m_k = (N_{cbps}/12)k_{mod(12)} + floor(k/12) \quad (1)$$

The second permutation is defined by [6],

$$j_k = s.floor(m_k/S) + [m_k + N_{cbps} - floor(12.m_k/N_{cbps})]_{mos(s)} \quad (2)$$

Where $k = 0, 1, \dots, N_{cbps}$; N_{cbps} is the number of coded bits per subcarrier, i.e., 1, 2, 4 or 6 for BPSK, QPSK, 16-QAM, or 64-QAM, respectively; k is the index of the coded bit before the first permutation; m_k is the index of that coded bit after the first and before the second permutation, and j_k is the index after the second permutation, just prior to modulation mapping. The receiver also does the reverse operation following the two step permutation using equations (3) and (4) respectively:

$$f_i = S.floor(j/S) + [j + floor(12.j/N_{cbps})]_{mod(s)} \quad (3)$$

$$S_j = (12.f_j - (N_{cbps}).floor(12.f_j/N_{cbps})); \quad j = 1, 2, \dots, N_{cbps} \quad (4)$$

In Reed-Muller Encoder [7] Only the first bit is used for in and output. m must be smaller than 31 and r must be smaller than m . Reed-Muller codes are listed as $RM(d,r)$, where d is the order of

the code, and r sets the length of code, $n = 2^r$. RM codes are related to binary functions on field $GF(2^r)$ over the elements $\{0,1\}$. $RM(1,r)$ codes are parity check codes of length $n = 2^r$, rate $R = \frac{r+1}{n}$ and minimum distance $d_{\min} = \frac{n}{2}$ [3].

OFDM block [3] generates OFDM symbols based on the parameters like *fft_length*, *occupied_tones*, and *cp_length* and *a type of modulation and etc* [8]. The transmitted signal voltage to antenna as a function of time during any OFDM symbol is defined as [3]

$$S(t) = \left(e^{j2\pi t f_c t} \sum_{k=-N_{used}/2}^{N_{used}/2} a_k \cdot e^{j2\pi k \Delta f (t-T_g)} \right) \quad (5)$$

where t is time, elapsed since the beginning of the subject OFDM symbol with $0 < t < T_s$, a_k is a complex number ; the data to be transmitted on the carrier whose frequency offset index is k , during the subject OFDM symbol. It assigns a point in a QAM constellation, T_g is guard time, T_s is OFDM symbol duration including guard time, Δf is carrier frequency spacing. In the subsequence, carriers are distinguished by a carrier index; however in order to reconstruct the OFDMA signal, frequency offset index is required. OFDMA is a special case or multi user version of OFDM which offers frequency diversity by spreading out the carriers all over the applied spectrum. Frequency offset index is defined in terms of its carrier index by equation (6)

$$k_{foi} = \begin{cases} k_{ci} - N_{used}/2, & k_{ci} < N_{used}/2. \\ k_{ci} - N_{used}/2 + 1, & k_{ci} \geq N_{used}/2. \end{cases} \quad (6)$$

Where K_{foi} is carrier frequency offset index, K_{ci} is carrier index and N is number of used carriers. Chunks to symbols block [3] maps a stream of symbol indexes (unpacked bytes or shorts) to stream of float or complex constellation points. Input is stream of short and output is stream of float.

$$out[nD+k] = symbol_table[in[n]D+k], k = 0, 1, \dots, D-1. \quad (7)$$

The combination of *gr_packed_to_unpacked_XX* followed by *gr_chunks_to_symbols_XY* deals the general case of mapping from a stream of bytes or shorts into arbitrary float or complex symbols.

Poly phase resample block [3] accepts a single complex stream in and outputs a single complex stream out. As such, it needs no extra glue to deal the input/output streams. This block is supplied to be consistent with the interface to the other PFB (Poly phase filter banks) block [3]. PFBs are a very powerful set of filtering tools that can efficiently perform many multi-rate signal processing tasks. GNU Radio has a set of PFBs to be employed in all sorts of applications. This block consents a signal stream and performs arbitrary resampling. The resampling rate can be any real number r . The resampling is acted by constructing N filters where N is the interpolation rate. Then D can be defined as, $D = \text{floor}(N/r)$. Using N and D , rational resampling is performed. where N/D is a rational number close to the input rate r where $i+1 = (i + D) \% N$. To acquire the arbitrary rate, interpolation between two points is required. For each value out, an output from the current filter, i , and the next filter $i+1$ are considered and then linearly interpolate between the two based on the real resampling rate.

Table 1. SNR estimation with variation in multiply const value.

Multiply Constant value	Estimated SNR Value
1	1.00234
10	10.6085
50	52.809883
100	108.647

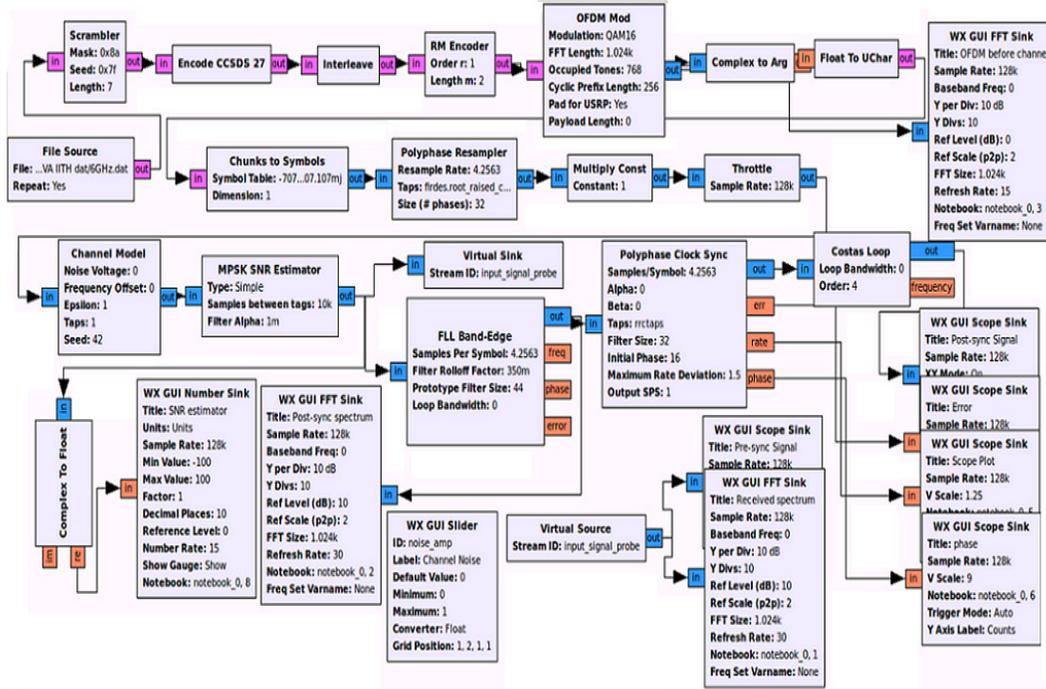


Figure 2: GNU schematic for OFDM signal transmission and reception over virtual source and sink.

Channel Model block [3] implements a basic channel model simulator that can be applied to help evaluate, design, and test various signals, waveforms, and algorithms. This model appropriates the user to set the voltage of an AWGN noise source, a (normalized) frequency offset, a sample timing offset, and a noise seed to randomize the AWGN noise source [9]. Multipath can be estimated in this model by using a FIR filter representation of a multipath delay profile. MPSK SNR estimator [3] is block for computing SNR of a signal. This block can be employed to monitor and retrieve estimations of the signal SNR. It is designed to work in a flow graph and passes all incoming data along to its output. Estimated SNR value is increased as multiply const block value increased (shown in Table 1) i.e. multiply const block acts as an amplifier in the schematic (See Fig. 2).

The frequency lock loop [3] derives a band-edge filter that covers the upper and lower bandwidths of a digitally modulated signal. The bandwidth range is determined by the excess bandwidth (e.g., roll off factor) [3] of the modulated signal. The placement in frequency of the band-edges is determined by the oversampling ratio (number of samples per symbol) and the excess bandwidth. The size of the filters should be fairly large so as to average over a number of symbols. The FIR filters are employed here because the filters have to have a flat phase response over the entire frequency range to allow their comparisons to be valid. It is very important that the band edge filters be the derivatives of the pulse shaping filter, and that they be linear phase. Otherwise, the variance of the error will be very large. Poly phase clock sync block performs

timing synchronization for PAM signals by minimizing the derivative of the filtered signal, which in turn maximizes the SNR and minimizes ISI [8].

The Costas loop [10] can have two output streams: stream 1 is the baseband I and Q; stream 2 is the normalized frequency of the loop. Digital Costas loop consists of Direct Digital Synthesizer (DDS), Low Pass Filter (LPF) and a Phase Discriminator (PD) and a Loop Filter (LF). Suppose that the input signal is a baseband signal modulated by the intermediate frequency carrier signal is given by [10]

$$x(t) = M(t).cos(w_c t) \quad (8)$$

The in phase and quadrature branch outputs of local DDS s are as follows respectively

$$V_{oi} = cos(w_c t + \Delta\phi) \quad (9)$$

$$V_{oq} = sin(w_c t + \Delta\phi) \quad (10)$$

Where $\Delta\phi$ is the phase difference between input signal and local signal of DDS. Then the multiplier outputs of in phase and quadrature branch are as follows

$$Z_i(t) = M(t).cosw_c t.cos(w_c t + \Delta\phi) \quad (11)$$

$$Z_q(t) = M(t).cosw_c t.sin(w_c t + \Delta\phi) \quad (12)$$

After low pass filtering, the corresponding outputs are

$$y_i(t) = \frac{1}{2}k_{l1}M(t)cos(\Delta\phi) \quad (13)$$

$$y_q(t) = \frac{1}{2}k_{l2}M(t)sin(\Delta\phi) \quad (14)$$

Where k_{l1} , k_{l2} are low pass filter coefficients. After $y_i(t)$ and $y_q(t)$ passed through phase discrimination and loop filter, following equation is obtained

$$V_c(t) = \frac{1}{8}k_p k_{l1} k_{l2} sin(2\Delta\phi) = k_d sin(2\Delta\phi) \quad (15)$$

4. CHANNEL CODING

In digital communications, a channel code is the term relating to the forward error correction code and interleaving in communication and storage where the communication media is recognized as a channel. The channel code is utilized to defend data sent over it for storage or recovery even in the orientation of noise (errors). Channel coding [6] is referred to process in both transmitter and receiver of a digital communications framework. Channel coding is made out of three techniques, for example Randomization, FEC (Forward Error Correction) and Interleaving.

5. EXPERIMENTAL RESULTS

A real time Software Defined Radio (SDR) is developed (as shown in Figure 3) by using a laptop with 8 Giga Bytes of RAM and an Intel® Core™ i5-3210M CPU clocked at 2.50 GHz. The integrated 1000Base-T Ethernet interface was connected to the USRPN210, equipped with the CBX daughterboard which is a full-duplex, wide band transceiver that extends a frequency band from 1.2 GHz to 6 GHz with a instantaneous bandwidth of 40 MHz (set up shown in Figure. 4). The CBX can serve a wide variety of application areas, including Wi-Fi research, cellular base stations, cognitive radio research, and RADAR. Required OFDM parameters for WiMAX specifications are mentioned in Table 1. Figure. 4 presents the encoding-decoding block diagram of the concatenated coding system for BER analysis [11] over air using USRP source and sink. RM coder (Reed-Muller code) [7] and CCSDS encoder (Convolutional coder) [3] are employed as inner code and outer code respectively. The concatenated OFDM signal is transmitted by USRP N210 RF front end by using TX/RX antenna and received by RX2 antenna over air (see Figure 3) in the lab environment. BER performance is analyzed by varying bits per symbol and window size in Error rate block and the results are observed in Table 3. It can be concluded that BER performance is improved as number of bits per symbol is increased and varies with window size. As modulation scheme size increases BER also increased (Observe Table 5) which is not desirable. Hence, while preferring a type of modulation scheme, various parameters have to be taken in to consideration.

In an OFDM transmission, we know that the transmission of cyclic prefix does not carry ‘extra’ information in Additive White Gaussian Noise channel. The signal energy is spread over time T_d+T_{cp} whereas the bit energy is spread over the time T_d i.e.

$$E_s(T_d + T_{cp}) = E_b.T_d$$

$$E_s = \frac{T_d}{T_d + T_{cp}}.E_b \quad (16)$$

The relation between symbol energy and the bit energy is given by

$$\frac{E_s}{N_0} = \frac{E_b}{N_0} \cdot \left(\frac{nDSC}{nFFT} \right) \left(\frac{T_d}{T_d + T_{cp}} \right) \quad (17)$$

Expressing in decibels

$$\frac{E_s}{N_0} dB = \frac{E_b}{N_0} dB + 10 \log_{10} \left(\frac{nDSC}{nFFT} \right) + 10 \log_{10} \left(\frac{T_d}{T_d + T_{cp}} \right) \quad (18)$$

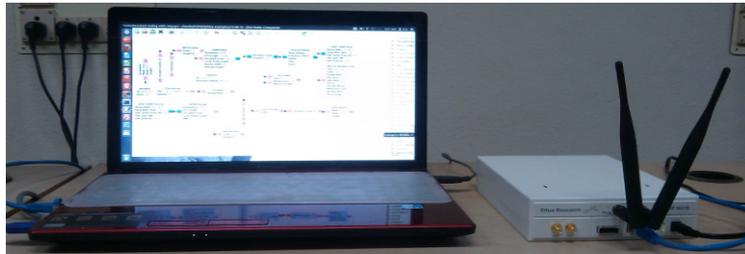


Figure 3: Software Defined Radio development test bed.

Table 2. Experimental parameters defined

Parameters	Values
FFT size (NFFT)	1024
Occupied Tones	768
Sampling rate	10.66667M
Center Frequency	2.48 GHz
Convolutional Code	1/2
Cyclic Prefix length	256
Useful symbol duration	91.43 μ s
Carrier spacing (1/Tu)	10.94 KHz
Guard time ($T_g=(1/4)* T_u$)	11.43 μ s
OFDM symbol duration	102.86 μ s
Mapping Schemes	BPSK, QPSK, 16QAM, 64QAM and 256QAM

Table 3. BER analysis with Bits per symbol and window size.

Bits per symbol	Window size	BER
1	10	3.5000000
1	1000	3.3710000525
1	10 ⁶	3.4379618168
4	10	0.8594650625
4	1000	0.84799997
4	10 ⁶	0.8594650625
8	10	0.4250000119
8	1000	0.42687499
8	10 ⁶	0.4296149015

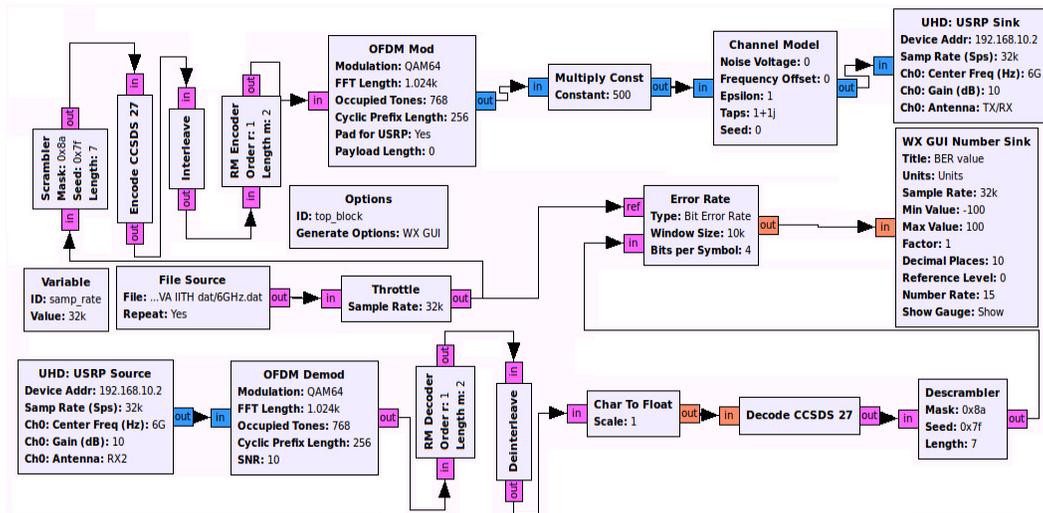


Figure 4: GNU Schematic for OFDM transmission and reception over USRP source and sink for BER analysis.

Table 4. BER and SNR analysis with Modulation scheme.

Modulation scheme	BER	SER
BPSK	0.8487750292	0.9564999938
QPSK	0.8616499901	0.9578999877
8PSK	0.8636000156	0.9592999816
16QAM	0.8638749719	0.9585999846
64QAM	0.8650249839	0.9577000141
256QAM	0.8660741590	0.9564999938

In order to evaluate the error probability, without loss of generality, paper focuses on the signal received on the first subcarrier, dropping the block index l for the sake of simplicity. A scaled version of the decision variable is given by

$$Z_1 = \lambda_1 Z_{EQ,1} = m_1 \lambda_1 S_1 + \sum_{n=2}^N m_n \lambda_n S_n + v_1 \quad (19)$$

Where $Z_{EQ,1} = Z_{EQ}[l]_1$, $S_n = S[l]_n$, $v_1 = \square[l]^{-1} v[l]_1$ and

$$m_n = [M]_{1,n} = \frac{\sin(\pi(n-1+\epsilon))}{N \sin \frac{\pi(n-1+\epsilon)}{N}} \cdot e^{j\pi \frac{N-1}{N}(n-1)} \quad (20)$$

represents the ICI coefficient due to the n^{th} subcarrier for $n=2, \dots, N$, and the attenuation factor of the useful data when $n=1$.

A possible approach to obtain BER (or equality, SER) consists of two steps. Firstly calculate the conditional bit error probability $P_{BE}(S, \lambda)$ that depends on the symbols in $S = [s_1, \dots, s_N]^T$ and on the channel amplitudes in $\lambda = [\lambda_1, \dots, \lambda_N]^T$. Successfully, $P_{BE}(S, \lambda)$ should be averaged over the joint probability density function (pdf) $f_{S, \Lambda}(S, \lambda) = f_S(S) f_{\Lambda}(\lambda)$ of the symbols and channel amplitudes is given by

$$BER = \int_{S, \lambda} f_S(S) f_{\Lambda}(\lambda) dS d\lambda \quad (21)$$

Figure. 2 shows the GRC schematic drawn for analysis of channel noise effect on concatenated OFDM signal over virtual sources and sink. Figures. 5, 6 & 7 present the OFDM signal, post synchronized spectrum and post synchronized signal before applying channel noise respectively. Figure 7 shows the various parameters such as channel noise, frequency offset, timing offset, timing beta, FLL (Frequency Lock Loop) bandwidth, Costas loop (phase) bandwidth and filter roll off factor which are effecting the transmitted signal. When one of these parameters is varied, received/synchronized signals are changed accordingly. Out of band radiation has become indistinguishable to the in band radiation due to channel noise (shown in Figure 8). Received and synchronized signals are disturbed by the channel noise and became glazed over (shown in Figures 9 & 10). Received resembled signal after Costas loop is shown in Figure 11. Post synchronized signal is effected by timing alpha and Costas loop bandwidth is shown in Figures 12 & 13. The phase variation in the signal is shown in Figure 14 with the effect of timing beta.

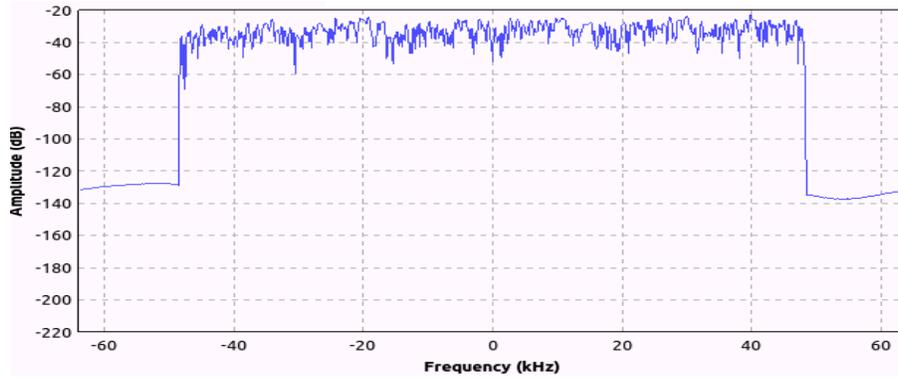


Figure 5: OFDM signal before passes through channel.

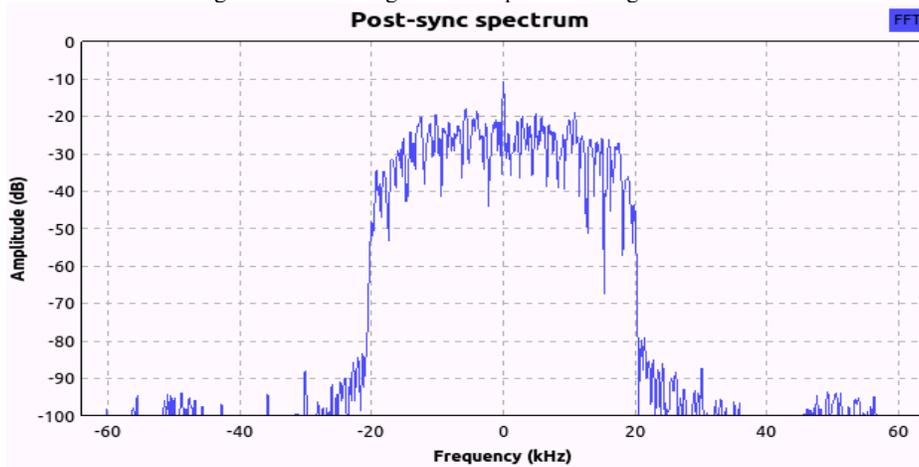


Figure 6: Post synchronized spectrum without channel effect.

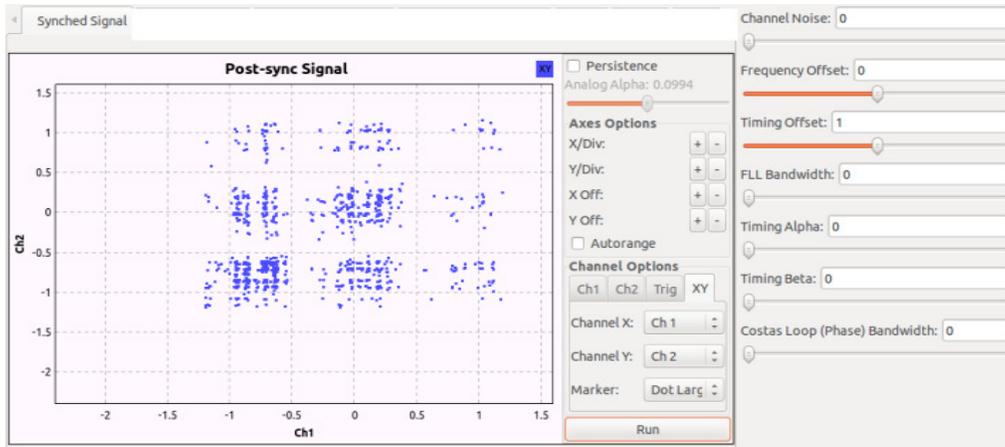


Figure 7: Post synchronized spectrum without channel effect.

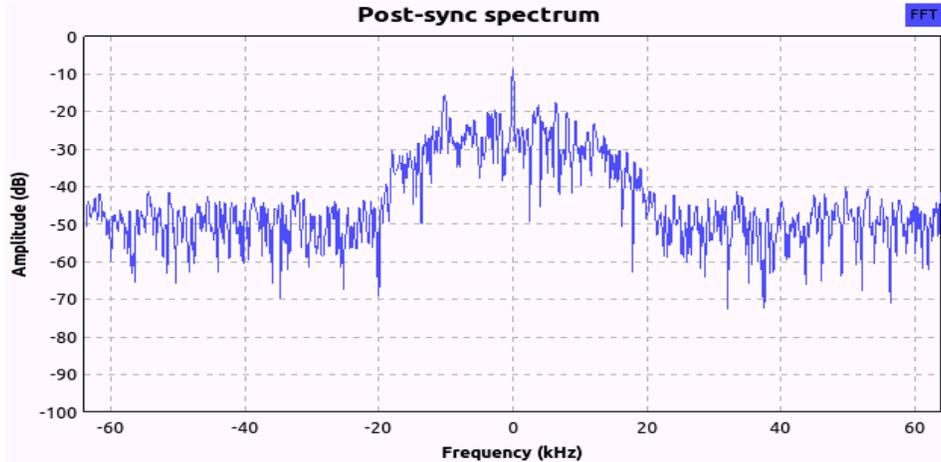


Figure 8: Post synchronized spectrum with effect of channel noise is 100m units.

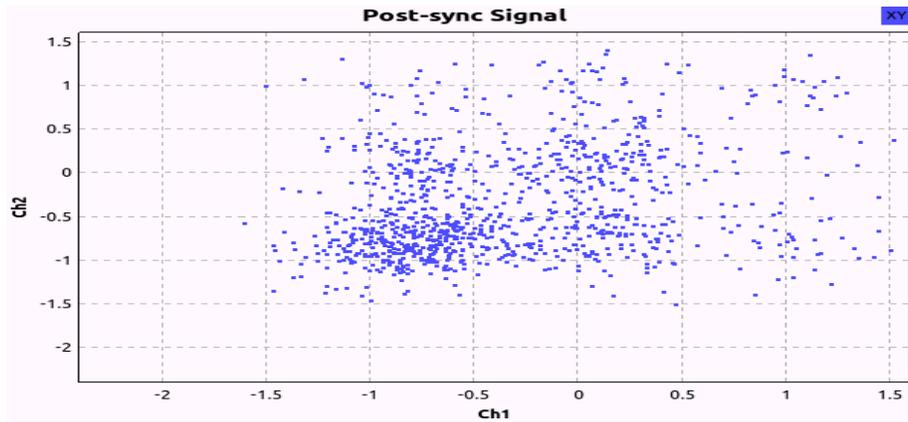


Figure 9: Post synchronized signal with effect of channel noise is 400m units.

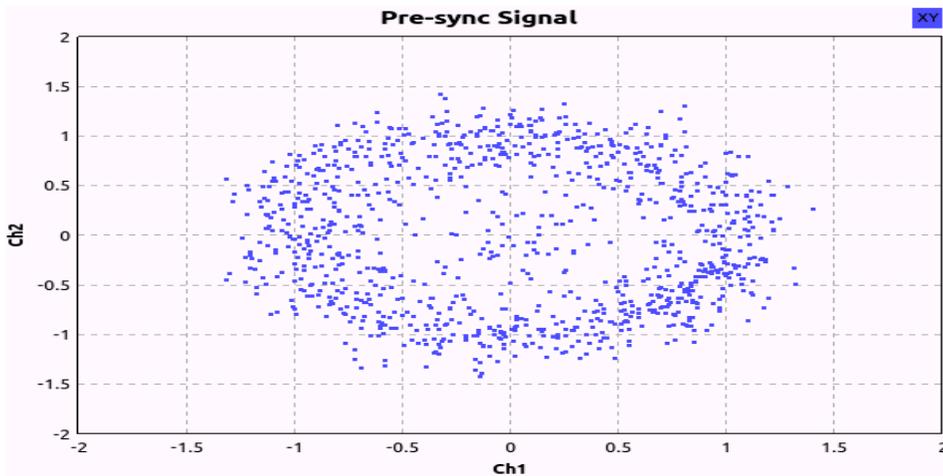


Figure 10: Received signal with effect of frequency offset is 11m units.

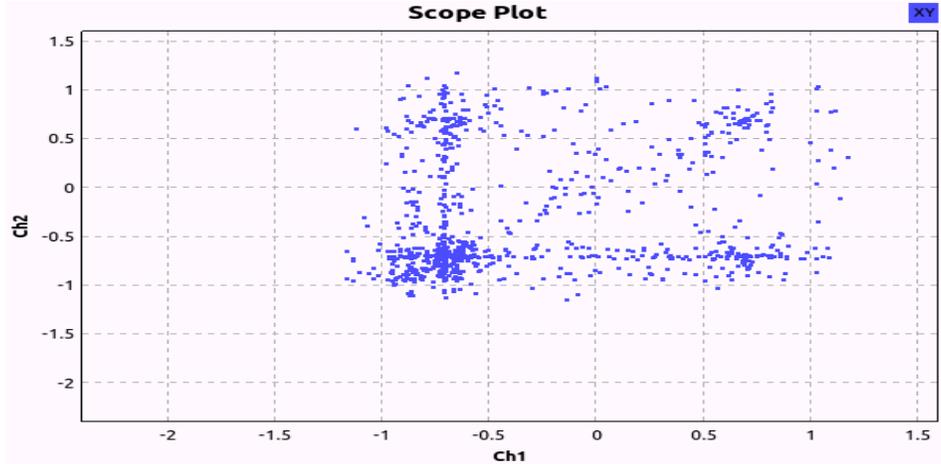


Figure 11: Received signal with effect of frequency offset is 11m units.

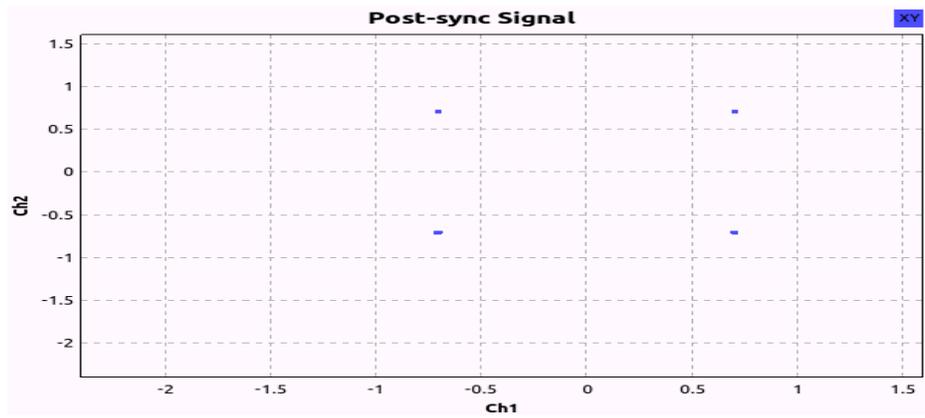


Figure 12: Post synchronized signal with effect of timing alpha is 20m units.

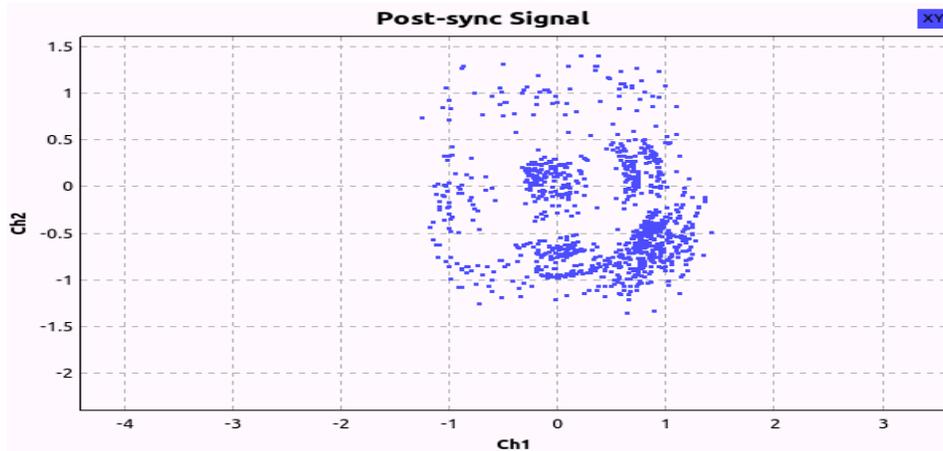


Figure 13: Post synchronized signal with effect of Costas loop (phase) bandwidth is 10m units.

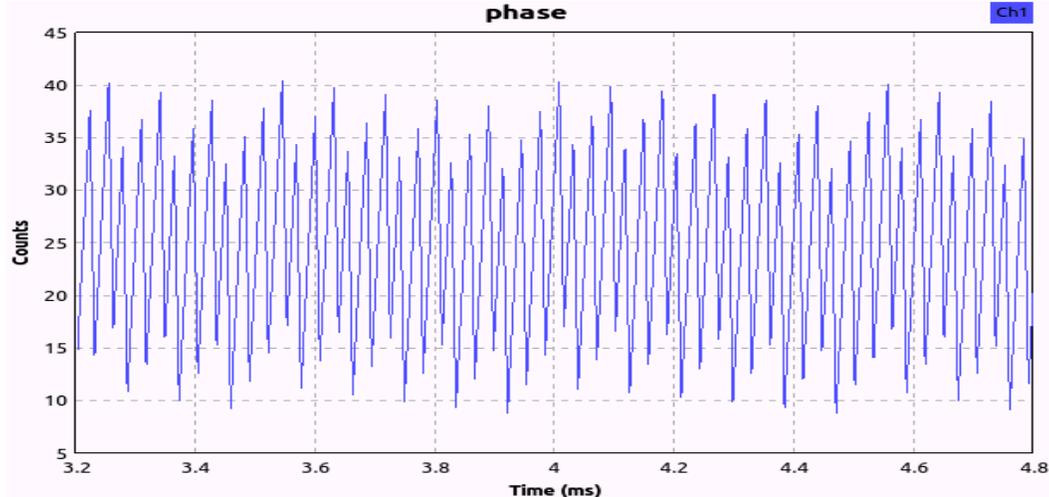


Figure 14: Phase variation with effect of timing beta is 10m units.

6. CONCLUSION

This paper shows the advancement of software defined radio (SDR) utilizing USRP N210 fittings and GNU Radio programming. OFDM/OFDMA based physical layer is executed with concatenated coding by considering RM code as internal code and Ccsds code as external code for Mobile WiMAX determinations at different (i) modulation schemes (ii) Channel noise levels (iii) frequency offsets (iv) costas loop bandwidth and (v) phase variation. As a result of the comparative study, it was found that: when channel conditions are poor, energy efficient schemes such as BPSK or QPSK were used and as the channel quality improves, 16-QAM or 64-QAM was used. It adjusts the modulation method almost instantaneously for optimum data transfer, thus making a most efficient use of the bandwidth and increasing the overall system capacity. Out of band radiation has gotten unclear to the in band radiation because of channel noise. Experiments validate the effectiveness of the proposed scheme in real time.

In this work, the measurement setup was somewhat idealized, since all measurements were conducted in a shielded environment. Future work should also include more realistic scenarios, such as interference from other secondary users or neighbouring frequency bands. More practical and better use of varying gain control should also be considered.

REFERENCES

- [1] Luiz Garcia Reis, A.; Barros, A.F.; Gusso Lenzi, K.; Pedroso Meloni, L.G.; Barbin, S.E., "Introduction to the Software-defined Radio Approach," *Latin America Transactions, IEEE (Revista IEEE America Latina)*, vol.10, no.1, pp.1156,1161, Jan. 2012.
- [2] Ettus, Matt. "USRP User's and Developer's Guide," Ettus Research LLC, 2005.
- [3] Radio, G. N. U. "The gnu software radio." Available from World Wide Web: <https://gnuradio.org>, 2007.
- [4] Andrews, Jeffrey G., Arunabha Ghosh, and Rias Muhamed, *Fundamentals of WiMAX: understanding broadband wireless networking*. Pearson Education, 2007.
- [5] Reddy, B.S.K.; Lakshmi, B., "Channel estimation and equalization in OFDM receiver for WiMAX with Rayleigh distribution," *Advanced Electronic Systems (ICAES), 2013 International Conference on*, pp.337,339, 21-23 Sept. 2013

- [6] Pin-Han Chen; Jian-Jia Weng; Chung-Hsuan Wang; Po-Ning Chen, "BCH Code Selection and Iterative Decoding for BCH and LDPC Concatenated Coding System," *Communications Letters, IEEE* , vol.17, no.5, pp.980,983, May 2013.
- [7] Reddy, Kumar, B. Siva, and B. Lakshmi. "Channel Coding and Clipping in OFDM for WiMAX using SDR." *International Journal on Recent Trends in Engineering & Technology*, 2013.
- [8] Reddy, B. Siva Kumar, and B. Lakshmi. "Adaptive Modulation and Coding in COFDM for WiMAX Using LMS Channel Estimator." , 2013.
- [9] Reddy, B.S.; Boppana, L.; Srivastava, A.; Kodali, R.K., "Modulation switching in OFDM for WiMAX through Rayleigh fading channel using GNU radio," *Advanced Electronic Systems (ICAES), 2013 International Conference on* , pp.331,333, 21-23 Sept. 2013
- [10] Guo, Si Yu, Dong Hai Qiao, and He Ming Zhao. "Implementation of Costas Loop for BPSK Receivers Using FPGA." *Applied Mechanics and Materials* 263, pp.990-993, 2013.
- [11] Rugini, Luca, and Paolo Banelli. "BER of OFDM systems impaired by carrier frequency offset in multipath fading channels." *Wireless Communications, IEEE Transactions on* 4.5: pp.2279-2288, 2005.

Authors

B. Siva Kumar Reddy, received B.Tech (E.C.E) and M.Tech (VLSI Design (Very Large Scale Integrated circuits Design)) degrees in from Jawaharlal Nehru Technological University, Hyderabad (JNTUH). Currently, he is working for doctorate in the field of wireless communications at National Institute of Technology, Warangal, India. He is an ISTE (Indian Society for Technical Education) life time member.



Dr. B. Lakshmi, has obtained B.Tech (E.C.E) from Nagarjuana university, M.Tech (EI) from NIT, Warangal, and Ph.D (VLSI Architectures) from I.I.T, Kharagpur. She is working as a faculty member in National Institute of Technology, Warangal since 1990. Her area of interests are Digital System Design, Microprocessor Systems and VLSI Architectures. She is reviewer for Elsevier journals in VLSI area.

