USE OF DVB-T AND DVB-S2 IN TELECARDIOLOGY

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WALEED ALSAADI

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I hereby declare that all information in this document has been obtained and presented in accordance with academic rules and ethical conduct. I also declare that, as required by these rules and conduct, I have fully cited and referenced all material and results that are not original to this work.

Name, last name: Waleed ALsaadi

Signature:

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ABSTRACT

Today's world offers technologies to make us communicate with each other in better ways than ever. It also enables data sharing to help us save people's lives. Telemedicine uses telecommunication and information technologies and provides clinical health care for people who are at a distance.

Telecardiology is type of telemedicine that uses this new communication system to send the electrocardiogram (ECG) signals and their timely transmission over a wireless network to remote healthcare professionals.

The aim of this thesis is to simulate the transmission of ECG signals over a communication system that uses digital video broadcasting-terrestrial (DVB-T) or digital video broadcasting – satellite version 2 (DVB-S2) technologies. bit error rate (BER) performance of this system is analyzed over additive gaussian white noise (AWGN) channel and compared to theoretical results. Keeping in mind that wireless channel suffers from multipath propagation, multiple-input multiple-output (MIMO) antenna technology is additionally used along with DVB-T.

It is shown that DVB-S2 technology offers performance improvements of up to 18 dB over DVB-T in an AWGN channel. It is also shown that using MIMO along with DVB-T mitigates the effects of multipath and improves the performance. This improvement is around 5 dB. Being a superior technology, however, does not necessarily mean DVB-S2 should be chosen over DVB-T in every circumstance. For example, in the case where less delay is important (i.e. in real-time transmission) DVB-T might still be the choice of transmission of ECG signals if performance degradation can be tolerated.

Keywords: Telemedicine, Telecardiology, DVB-S2, DVB-T, MIMO, ECG signals

ÖZET

Bugünün dünyası her zamankinden daha iyi bir şekilde iletişim yapmamız için bize teknolojiler sunuyor. Ayrıca bize insanların hayatını kurtarmak için veri paylaşımı sağlıyor. Teletıp telekomünikasyon ve bilişim teknolojilerini kullanır ve insanlara uzaktan sağlık bakımı vermemizi sağlar.

Telekardiyoloji bu yeni iletişim sistemini kullanan ve uzakta olan sağlık profesyonelleri için bir kablosuz ağ üzerinden elektrokardiyogram (EKG) sinyalleri ve onların zamanında iletimini sağlayan Teletıp türüdür.

Bu tezin amacı, dijital video yayıncılığı-karasal (DVB-T) veya dijital video yayını -uydu sürüm 2 (DVB-S2) teknolojilerini kullanan bir iletişim sistemi üzerinden EKG sinyallerinin iletimini simüle etmektir. Bu sistemin bit hata oranı (BER) performansı toplanır beyaz Gauss gürültüsü (AWGN) kanalı üzerinde analiz edilir ve teorik sonuçlar ile karşılaştırılır. Kablosuz kanallarda çokyollu yayılıma olduğunu için çoklu giriş çoklu çıkış (MIMO) anten teknolojisi ayrıca DVB-T ile birlikte kullanılır.

DVB-S2 teknolojisi AWGN kanalında DVB-T den 18 dB ye kadar performans artışı sağlamıştır. Aynı zamanda, DVB-T ile birlikte MIMO kullanarak çokyollu yayılımın etkileri azaltılabilir ve performansı artırışı sağlanabilir. Bu artış yaklaşık 5 dB dir. Üstün teknoloji olması ancak DVB-S2 nin DVB-T ile karşılaştırıldığında her durumda tercih edileceği anlamına gelmez. Örneğin, daha az gecikmenin önemli olduğu bir durumda (örneğin, gerçek zamanlı iletim), performans düşüşü tolere edilebilir eğer DVB-T hala EKG sinyallerinin iletimi seçimi olabilir.

Anahtar Kelimeler : Teletıp, telekardiyoloji, dijital video yayını -uydu sürüm 2 (DVB-S2), dijital video yayıncılığı-karasal (DVB-T), çoklu giriş çoklu çıkış (MIMO), EKG sinyalleri

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B	Transmission bandwidth (hertz)
C	Channel capacity (bits/s)
$c_n(t)$	The tap coefficients
$c_r(t)$ and $c_i(t)$	Gaussian with zero mean values
E_b/N_0	Energy per bit to noise power spectral density ratio
$f(\alpha)$	PDF of Rayleigh fading signal amplitude
FC	Carrier frequency
F_D	Doppler frequency associated with Rayleigh fading channels
F_M	Maximum Doppler frequency
K_{BCH}	Number of bits of BCH encoded Block
K_{LDPC}	Number of bits of LDPC encoded Block
M	Number of OFDM symbols
N	Number of sinusoids in Jakes' fading simulator
N_{LDBC}	Number of bits of LDPC coded Block
N_0	Single-sided noise power spectral density (watts/hertz)
N	Code length
$n_{k,t}$	zero mean Gaussian noise with variance $N_0/2$
P	Received signal power (watts)
$P(c_i\|y_i)$	Probability value for given input y_i
R	Code rate
1/W	Time resolution
A	Normalized Rayleigh fading factor

$\alpha(t)$	Rayleigh fading signal amplitude
ADC	Analog Digital Converter
AWGN	Additive White Gaussian Noise
BBFRAME	The set of K_{BCH} bits which form the input to one FEC encoding process
BCH	Bose- Chaudhuri- Hochquenghem multiple error code
BER	Bit Error Rate
Bps	Bit per second
CP	Cyclic Prefix (copy of the last part of OFDM symbol)
COFDM	Coded Orthogonal frequency Division Multiplexing
DMT	Discrete Multitude
DSNG	Digital Satellite News Gathering
DVB	Digital Video Broadcasting project
DVB-S	Digital Video Broadcasting- Satellite
DVB-S2	Second generation Digital Video Broadcasting-Satellite
DVB-T	Digital Video Broadcasting- Terrestrial specified in EN 300 421
DVB-T2	Second generation Digital Video Broadcasting-Terrestrial
ETSI	European Telecommunications Standards Institute
FDX	Full Duplex (communication channel)
FEC	Forward error correction
FEC FRAME	The set of N_{ldpc} (16200 or 64800) bits from one LDPC encoding operation.
FFT	Fast Fourier Transform
HDX	Half Duplex (communication channel)
ICI	Inter Carrier Interference

IFFT	Inverse Fourier Transform
ISDN	Integrated Services Digital Network
ISI	Inter Symbol Interference
ITU	International Telecommunications Union
LDPC	Low Density Parity Check (codes)
MCM	Multi Carrier Modulation
MPEG-2 TS	Moving picture Experts Group –ver2 transport stream
OFDM	Orthogonal Frequency- Division Multiplexing
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RMS	Root Mean Square
RS	Reed Solomon
RS-CC	Reed Solomon- Convolution Code
RSK	Rotation Shift Keying
SFN	Single Frequency Network
SNR	Signal-to-noise Ratio

CHAPTER 1
INTRODUCTION

1.1 Introduction

A huge number of individuals pass on every year from illnesses; the elderly are more vulnerable to such illnesses. Numerous retirement homes are introducing frameworks that can constantly and remotely monitor the electrocardiograms (ECGs) of their residents. For instance, Alarm Net (Ubeyli, 2008) is a helped living and private checking system that opens up new doors for nonstop observing of the elderly and those needing restorative help. Wearable ECG sensors can remotely monitor a patient's pulse, alarming medical staff to changes in status (Ghaffari et al., 2007). There are two issues related to data transmission of ECGs:

I. The information from a 12-lead ECG with 11-bit for one day of 300 Hz signal will be about 500 MB. Transmitting these amounts of information wirelessly will need high speed networks.
II. High Channel Error Rate. Remote channels are normally much noisier than wired connections and suffer from both multipath fading and shadowing, which can have a shocking effect on the apparent nature of a reproduced ECG signal.

Notwithstanding, to make ECG checking effective in retirement homes, we should have the capacity to monitor a few patients progressively.

1.2 The Aim of the Thesis

We aim in this thesis to simulate and analyze the use of Digital Video Broadcasting (DVB) technology in health care application special for Telecardiology. Specifically, DVB-T, a terrestrial standard, and DVB-S2, a satellite standard version 2, are utilized for the digital video broadcasting system. Simulations are carried out in additive white gaussian noise (AWGN) channel and compared with theoretical. Additionally, multiple-input multiple-output antenna technology is used in DVB-T in order to lessen the effects of terrestrial multipath fading.

1.3 Thesis Structure

The rest of this thesis is divided into 6 chapters and organized as follows:

Chapter 2 provides a detailed explanation on system model.

Chapter 3 introduces telemedicine, Telecardiology and ECG. This chapter also includes general block diagram of ECG signal measurement system, Telecardiology system block diagram, telemedicine system block diagram and their short explanations.

Chapter 4 discusses digital video broadcasting standard in detail.

Chapter 5 is about simulations carried out and their analysis.

Chapter 6 gives the conclusions.

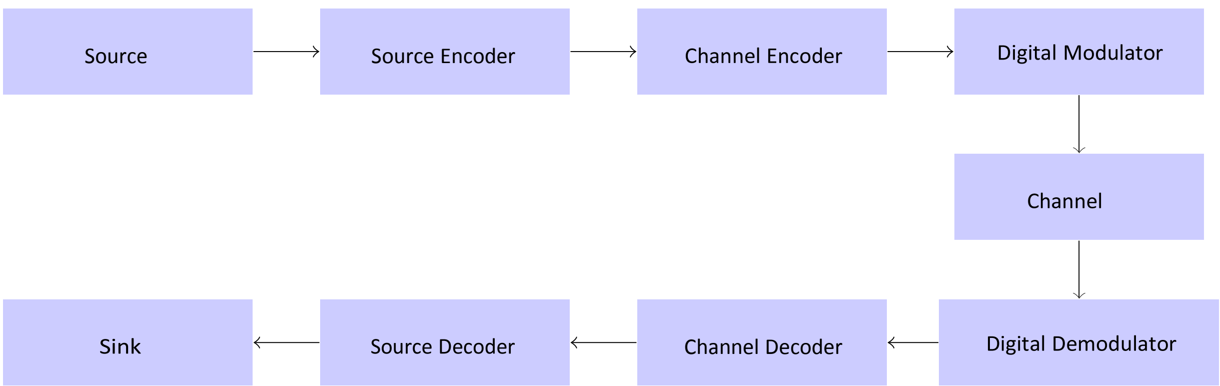
CHAPTER 2
SYSTEM MODEL

2.1 Source Encoder

The process by which information symbols are mapped to alphabetical symbols is called source coding. The mapping is generally performed in sequences or groups of information and alphabetical symbols. Also, it must be performed in such a manner that it guarantees the exact recovery of the information symbol back from the alphabetical symbols otherwise it will destroy the basic theme of the source coding. The source coding is called lossless compression if the information symbols are exactly recovered from the alphabetical symbols (Clifford et al., 2006).

Otherwise it is called lossy compression. Compression or bit-rate reduction process is the process of removing redundancy from the source symbols, which essentially reduces data size. Source coding is a vital part of any communication system as it helps to use disk space and transmission bandwidth efficiently. In case of lossless encoding, error free reconstruction of source symbols is possible, whereas, exact reconstruction of source symbols is not possible in case of lossy encoding. Minimum average length of codewords as a function of entropy is restricted between an upper and lower bound of the information source symbols by the Shannon's source coding theorem. For a lossless case, entropy is the maximum limit on the data compression, which is represented by H(x) and is defined as

$$\mathrm{H}(\mathrm{x}) = \sum_{\mathrm{i}=1}^{M} \mathrm{p_x}(\mathrm{i}) * \log \frac{1}{\mathrm{p_x}(\mathrm{i})} \tag{2.1}$$



where $p_x(i)$ is the probability distribution of the information source symbols and M is the number of codes .

The Huffman algorithm is basically used for encoding entropy and to compress data without loss. In order to choose a particular representation for each symbol, Huffman coding makes use of a particular method that leads to a prefix code also called as prefix-free code. This method uses a minimum number of bits in the form of strings to represent mostly used and common source symbols and vice versa. Furthermore, Huffman coding uses a code table of varying length in order to encode a source symbol. The table is made on the basis of the probability calculated for all the values of the source symbol that have the possibility to occur. This scheme is optimal in terms of obtaining codewords of minimum yet possible average length (Camm et al., 2009).

The coding concept of Shannon-Fano is used to construct prefix code using a group of symbols along with the probabilities by which they are measured or calculated. This technique, however, is not the optimum as it does not guarantee the codeword of minimum yet possible average length as does the Huffman coding. In contrast, this coding technique always achieves code words lengths that lie within one bit range of the theoretical and ideal log p(x).

Shannon-Fano-Elias coding, on the other hand, works on cumulative probability distribution. It is a precursor to arithmetic coding, in which probabilities are used to determine codewords. The Shannon-Fano-Elias code is not an optimal code if one symbol is encoded at time and is a little worse than, for instance, a Huffman code. One of the limitations of these coding techniques is that all the methods require the probability density function or cumulative density function of the source symbols which is not possible in many practical applications. The universal coding techniques are proposed to address this dilemma, for instance, Lempel-Ziv (LZ) coding and Lempel-Ziv-Welch (LZW) coding. The LZ coding algorithm is a variable-to-fixed length, lossless coding method. A high level description of the LZ encoding algorithm is given as (Luna, 2008):

1. Initializing the dictionary in order to have strings/blocks of unity length.

2. Searching the lengthiest block W in the dictionary that matches the current input string.

3. Encode W by the dictionary index and remove W from the input.

4. Perform addition of W which is then followed by the next symbol as input to the dictionary.

5. Move back to step 2.

The LZ compression algorithm shows that this technique asymptotically achieves the Shannon limit.

The rate distortion theory highlights and discusses the source coding with losses, i.e., data compression. It works by finding the minimum entropy (or information) R for a communication channel in order to approximately reconstruct the input signal at the receiver while keeping a certain limit of distortion D. This theory actually defines the theoretical limits for achieving compression using techniques of compression with loss. Today's compression methods make use of transformation, quantization, and bit-rate allocation techniques to deal with the rate distortion functions. The founder of this theory was C. E. Shannon (Luna, 2008). The number of bits that are used for storing or transmitting a data sample are generally meant as rate. The Mean Squared Error (MSE) is a commonly used distortion measure in the rate-distortion theory. Most of lossy compression techniques operate on data that will be perceived by human consumers (listening to music, watching pictures and video), therefore, the distortion measure should be modeled on human perception. For example, in audio compression, perceptual models are comparatively very promising and are commonly deployed in compression methods like MP3 or Vorbis. However, they are not convenient to be taken into account in the rate distortion theory. Also, methods that are perception dependent are not considered promising for image and video compression so they are in general used with Joint Picture Expert Group (JPEG) and Moving Picture Expert Group (MPEG) weighting matrix.

Some other methods such as Adam7 Algorithm, Adaptive Huffman, Arithmetic Coding, Canonical Huffman Code, Fibonacci Coding, Golomb Coding, Negafibonacci Coding, Truncated Binary Encoding, etc., are also used in different applications for data compression (Purvis et al., 1999).

2.2 Channel Encoder

The channel coding is a framework of increasing reliability of data transmission at the cost of reduction in information rate. This goal is achieved by adding redundancy to the information symbol vector resulting in a longer coded vector of symbols that are distinguishable at the output of the channel (Kabir et al., 2008).

Channel coding methods can be classified into the following two main categories:

1. Linear block coding maps a block of k information bits onto a codeword of n bits such that n > k. In addition, mapping of k information bits is distinct, that is, for each sequence of k information bits, there is a distinct codeword of n bits. Examples of linear block codes include Hamming Codes, Reed Muller Codes, Reed Solomon Codes, Cyclic Codes, BCH Codes, etc. A Cyclic Redundancy Check (CRC) code can detect any error burst up to the length of the CRC code itself.

2. Convolutional coding maps a sequence of k bits of information onto a codeword of n bits by exploiting knowledge of the present k information bits as well as the previous information bits. Examples include Viterbi-decoded convolutional codes, turbo codes, etc.

The major theme of channel coding is to allow the decoder to decode the valid codeword or codeword with noise which means some bits would be corrupted. In ideal situation, the decoder knows the codeword that is sent even after corruption by noise. C. E. Shannon in his landmark work proposed a framework of coding the information to be transmitted over a noisy channel. He also provided theoretical bounds for reliable communication over a noisy channel as a function of the channel capacity (Reimers, 1996).

2.3 Channel Modeling

The channel is defined as a single path for transmitting signals in either one path only HDX or in both paths FDX. The goal of wireless channel modeling is to find useful analytical models for the variations in the channel. The most noticeable problem of the wireless communications is channel fading. Various properties such as multipath propagation, terminal mobility and user interference, result in channel with time-varying parameters. Fading of the wireless channel can be classified into large-scale and small-scale fading. Large-scale fading includes the variation of the mean of the received signal power over large distances relative to the signal wavelength (Jakes et al., 1994). On the other hand, small-scale fading includes the modulation and demodulation schemes that are tough to these variations. Hence we focus on the small scale variations in this class. Reflection, diffraction and scattering in the communication channel causes fast variations in the received signal. The reflected signals arrive at different delays which cause random amplitude and phase of the received signals. This singularity is called multipath fading. If the product of the root mean square (RMS) delay spread (standard deviation of the delay spread) and the signal bandwidth is much less than unity, the channel is said to suffer from fading. The relative motion between the transmitter and the receiver (or vice versa) causes the frequency of the received signal to be changed relative to that of the transmitted signal. The frequency change, or Doppler frequency, is proportional to the velocity of the receiver and the frequency of the transmitted signal. A signal suffers slow fading when the bandwidth of the signal is much larger than the Doppler spread (defined as a measure of the spectral broadening caused by the Doppler frequency). The combination of the multipath fading with its time variations causes the received signal to degrade severely. This degradation of the quality of the received signal caused by fading needs to be compensated by various techniques such as diversity and channel coding (Jakes et al., 1994).

2.3.1. AWGN channel

Additive white Gaussian noise (AWGN) is a channel model which can be expressed as linear addition of wideband or white noise with a constant spectral density and an amplitude of Gaussian distribution (Jakes et al., 1994). Any wireless system in AWGN channel can be expressed as $y = x + n$, where n is the additive white Gaussian noise, x and y are the input and output signals in turn. The AWGN channel model does not account for fading, frequency selectivity or dispersion. The source of Gaussian noise comes from many ordinary sources such as thermal vibrations of atoms in antennas, shot noise, black body radiation from the warm objects and etc. However this channel is very useful model for many satellite and deep space communication connections. The AWGN channel can be demonstrated as in Figure 2.2 Channel capacity formula is a function of channel characteristics such as received signal and noise powers. As a matter of fact a number of different formulas are commonly used for calculating channel capacity (ETSI, 2008). For Shannon equation the channel capacity can be expressed as in (2.2).

$$C = B * \log_2(1 + \frac{P}{N_0 B}) \qquad (2.2)$$

where,

C=channel capacity (bits/s)

B=transmission bandwidth (hertz)

P=received signal power (watts)

N_0= single-sided noise power spectral density (watts/hertz)

2.3.2. Rayleigh fading channel

The Rayleigh fading channel, usually mentioned as a worst-case fading channel is a statistical model for the effect of a propagation environment on a radio signal, such as that used by wireless devices (Rayleigh Fading, 2011). It assumes that the magnitude of a signal that has passed through such a transmission medium (also called a communications channel) will vary randomly, or fade, according to a Rayleigh distribution Received signal can be modeled as $y = \alpha * t_e + n$. Here, α is the normalized Rayleigh fading factor related to the fading coefficient of the channel c(t) through $\alpha = |c(t)|$, where the real and imaginary components of c(t) are Gaussian random variables. If sufficient channel interleaving is presented, then fading coefficients of c(t) are independent. Rayleigh fading is seen as a reasonable model for heavily built-up urban environments on radio signals (ETSI, 2011). Rayleigh fading is most appropriate when there is no major propagation along a line of sight between the transmitter and the receiver. If there is a line of sight, Rician fading may be more applicable. A general model for time-variant multipath channel is shown in Figure 2.1. The channel model consists of a tapped delay line with uniformly spaced taps. The tap spacing is 1/W, where W is the amount of signal transmitted through the channel as FIR filter.

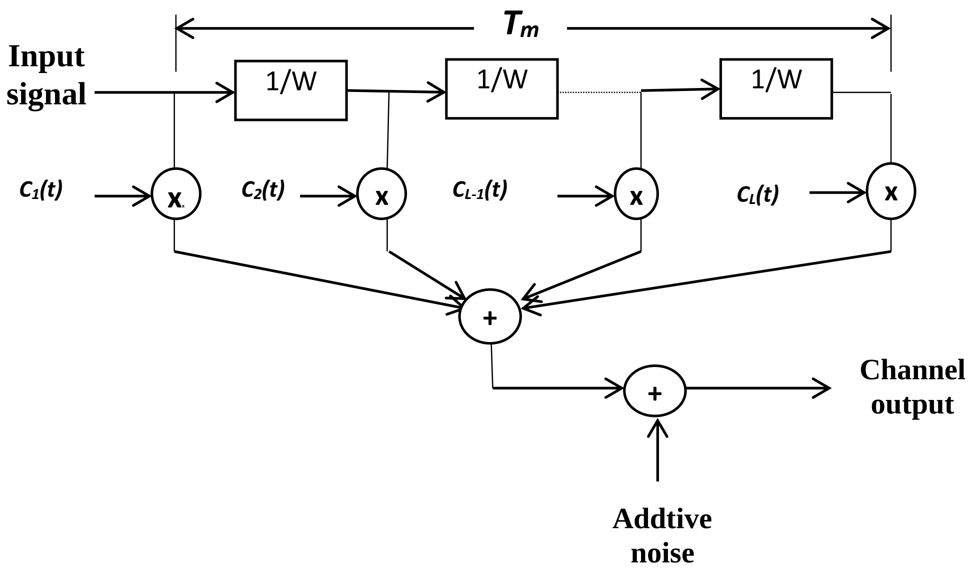


Figure 2.1: Model for time-invariant multipath channel

As a result 1/W is the time resolution that can possibly be achieved by transmitting a signal with bandwidth W. The tap coefficients are denoted as $c_n(t) \equiv \alpha_n(t)*\exp(j*\varphi_n(t))$ are usually modeled as complex valued, Gaussian random processes (ETSI, 2009). Each of the tap coefficients can be expressed as

$$c(t) = c_r(t) + jc_i(t) \tag{2.3}$$

$$c(t) = \alpha_t e^{j\varphi(t)} \tag{2.4}$$

where

$$\alpha(t) = \sqrt{c_r^2(t) + c_i^2(t)} \tag{2.5}$$

$$\varphi(t) = tan^{-1}\frac{c_i(t)}{c_r(t)} \tag{2.6}$$

In this representation $c_r(t)$ and $c_i(t)$are Gaussian with zero-mean values, the amplitude $\alpha(t)$ is characterized statistically by the Rayleigh probability distribution and $\varphi(t)$is independent random variable which is uniform on [0, 2π].

The Rayleigh fading signal amplitude is described a

$$f(\alpha) = \frac{\alpha}{\sigma^2} e^{\frac{\alpha^2}{2\sigma^2}},\ \alpha \geq 0 \qquad (2.7)$$

2.3.3. Rician fading channel

When there is line-of-sight, direct path is normally the strongest component among the other reflections from other paths which all goes into deeper fade compared to the multipath components. This kind of signal is approximated by Rician distribution. As the dominating component run into more fade the signal characteristic goes from Rician to Rayleigh distribution (DVB, 2008). The derivation of the probability density function of the amplitude is more involved than for Rayleigh fading, and a Bessel function occurs in the mathematical expression. In the presence of such a path, the transmitted signal can be written as:

$$s(t) = \sum_{i=1}^{n-1} a_i \cos(\omega_c t + \omega_{di} t + \varphi_i) + K_d \cos(\omega_c t + \omega_d t) \qquad (2.8)$$

where the constant K_d is the strength of the direct component, ω_d is the Doppler shift along the line-of-sight path, and ω_{di} are the Doppler shifts along the indirect paths and *n* is the total number of paths. The envelope in this case has a Rician density function given by:

$$f(r) = \frac{r}{\sigma^2} e^{\frac{-\{r^2 + K_d{}^2\}}{2\sigma^2}} . I_0 \left\{ \frac{r K_d}{\sigma^2} \right\} \qquad (2.9)$$

2.4 OFDM-based Wireless Communication Systems

Orthogonal frequency-division multiplexing (OFDM), is also known as multicarrier modulation (MCM) or discrete multitone (DMT) is a famous modulation technique that is tolerant to channel disturbances and impulse noise. Multi carrier modulation have been developed 1950's by introducing two modems, the Collins Kineplex system (ETSI, 2009) and the one so called Kathryn modem (ETSI, 2011) OFDM has extraordinary properties such as bandwidth efficiently, highly flexible in terms of its adaptability to channels and robustness to multipath. OFDM is used in many applications including high data rate transmission over twisted pair lines and fiber, digital video broadcasting terrestrial (DVBT), personal communications services and etc.

2.4.1. OFDM

To achieve higher spectral efficiency in multicarrier system, the sub-carriers must have overlapping transmit spectra but at the same time they need to be orthogonal to avoid complex separation and processing at the receiving end (Engels, 2002). As it is stated in (Engels, 2002), the orthogonal set can be represented as such:

$$\psi\ (t) = \{\frac{1}{\sqrt{Ts}} exp^{jwkt}\ for\ t \in [0, Ts]\} \tag{2.10}$$

with

$$w_k = w_0 + \mathrm{k}w_s;\ k=0,1,\ldots..N_c\text{-}1 \tag{2.11}$$

where w_0 is the lowest frequency used and w_k is the subcarrier frequency and w_s is the subcarrier spacing . Multicarrier modulation schemes that fulfill above mentioned conditions are called orthogonal frequency division multiplex (OFDM) systems. Instead of baseband modulator and bank of matched filters, Inverse Fast Fourier Transform (IFFT) and Fast Fourier Transform (FFT) is efficient method of OFDM system implementation as shown in Figure 2.2 since it is cheap and does not suffer from inaccuracies in analog oscillators. Inter-symbol interference (ISI) occurs when the signal passes through the time dispersive channel. In an OFDM system, it is also possible that orthogonality of the subscribers may be lost, resulting in inter-carrier interference (ICI). OFDM system uses cyclic prefix (CP) to overcome these problems. A cyclic prefix is the copy of the last part of the OFDM symbol to the beginning of transmitted symbol and is removed at the receiver before demodulation. The cyclic prefix should be at least as long as the length of impulse response (Wood et al., 2008). The use of prefix has two advantages: it serves as guard space between successive symbols to avoid ISI and it converts linear convolution with channel impulse response to circular convolution.

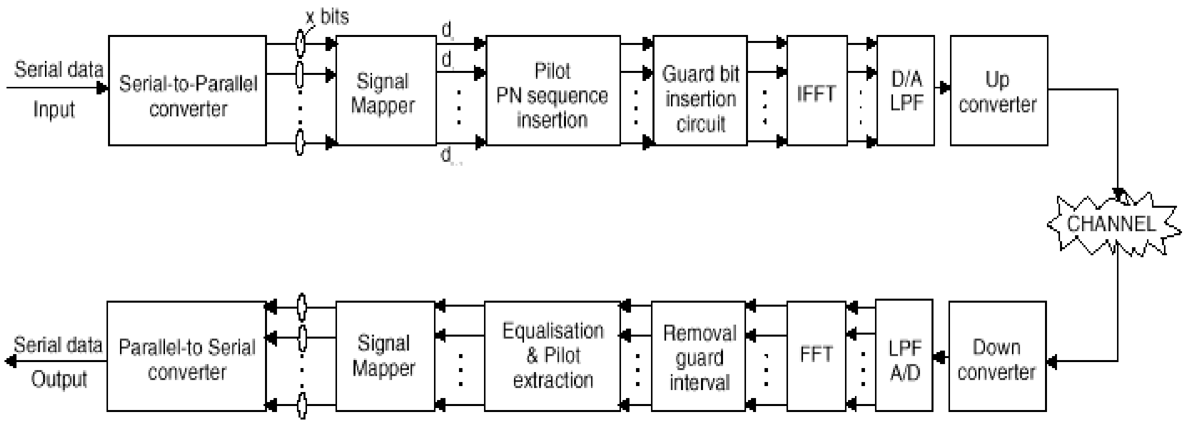


Figure 2.2: Model of OFDM system

As circular convolution in time domain translates into scalar multiplication in frequency domain, the subcarrier remains orthogonal. Moreover, there is no ICI. In Figure 2.2, L coded vector x_i are generated by proper coding, interleaving and mapping. After adding cyclic prefix, OFDM signal is passed through multipath channel. At the receiver the cyclic prefix is removed and received signal is passed through FFT block to get L received vectors y_i; where $n_{k,t}$ are zero mean Gaussian noise with variance $N_0/2$ of k th sample of the t th OFDM symbol. N_0 is the noise power, k =(1,2,...,$NFFT$ −1) and t =(1,2,...,M), where M is the number of OFDM symbols and NFFT is the size of FFT.

2.4.2 Cyclic prefix

Inter-symbol interference occurs when the signal passes through the time dispersive channel. In an OFDM system, it is also possible that orthogonality of the subscribers may be lost, resulting in inter carrier interference. OFDM system uses cyclic prefix (CP) to overcome these problems. A cyclic prefix is the copy of the last part of the OFDM symbol to the beginning of transmitted symbol and removed at the receiver before demodulation. The cyclic prefix should be at least as long as the length of impulse response. However, there is a limit on energy while increasing the length of cyclic prefix. As it is expected the energy increases as the cyclic prefix length increases. As it is expressed in (Chang, 1970) the SNR loss due to the usage of cyclic prefix can be evaluated using equation 2.12.

$$SNR_{loss} = -10\log_{10}(1 - \frac{T_{cp}}{T}) \tag{2.12}$$

In equation 2.12 T_{cp} refers to the cyclic prefix length. We can express the length of the transmitted symbol $T = T_{cp} + T_s$. Choosing the length of the cyclic prefix must be done carefully as seen in Figure 2.3. The following matters should be considered,

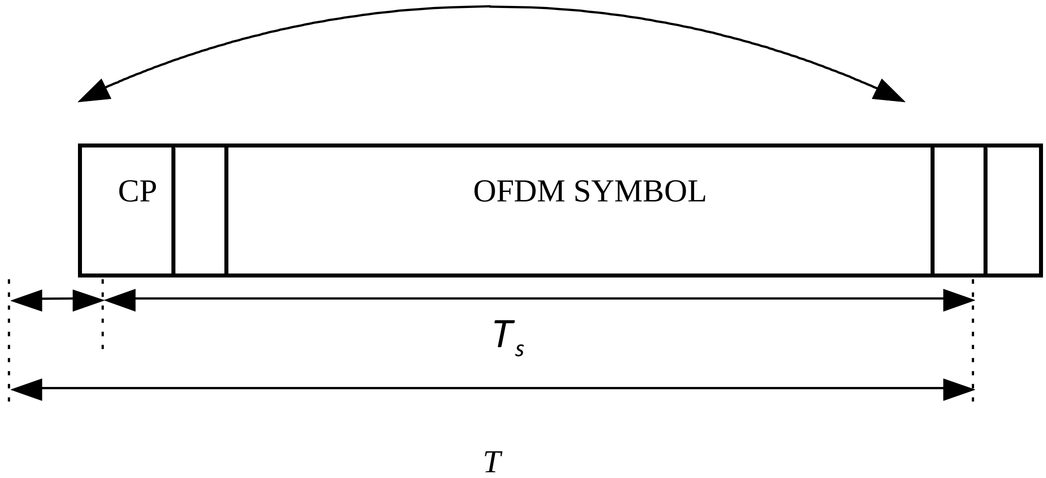


Figure 2.3: Cyclic Prefix

1. Number of symbols per second decreases to $R(1 - T_{cp}/T)$
2. The ratio T_{cp}/T must be kept as small as possible

As it is stated in [11] the width of the guard interval can be $R = 1/32$, $R = 1/16$, $R = 1/8$, or $R = 1/4$ that of the original block length. In our simulation we are using a guard interval width $R = 1/4$ of the original block length.

2.5 Multiple Input Multiple Output (MIMO)

Multiple-input multiple-output, or MIMO, is a radio communications technology or RF technology that is being mentioned and used in many new technologies these days. Wi-Fi, LTE (Long Term Evolution), and many other radio, wireless and RF technologies are using the new MIMO wireless technology to provide increased link capacity and spectral efficiency combined with improved link reliability using what were previously seen as interference paths. Even now there are many MIMO wireless routers on the market, and as this RF technology is becoming more widespread, more MIMO routers and other items of wireless MIMO equipment will be seen. As the technology is complex many engineers are asking what is MIMO and how does it work (Chang, 1970).

2.5.1 MIMO development and history

MIMO technology has been developed over many years. Not only did the basic MIMO concepts need to be formulated, but in addition to this, new technologies needed to be developed to enable MIMO to be fully implemented. New levels of processing were needed to allow some of the features of spatial multiplexing as well as to utilize some of the gains of spatial diversity (Fulford et al., 2004).

Up until the 1990s, spatial diversity was often limited to systems that switched between two antennas or combined the signals to provide the best signal. Also various forms of beam switching were implemented, but in view of the levels of processing involved and the degrees of processing available, the systems were generally relatively limited.

However with the additional levels of processing power that started to become available, it was possible to utilize both spatial diversity and full spatial multiplexing.

The initial work on MIMO systems focused on basic spatial diversity - here the MIMO system was used to limit the degradation caused by multipath propagation. However this was only the first step as system then started to utilize the multipath propagation to advantage, turning the additional signal paths into what might effectively be considered as additional channels to carry additional data (Chang, 1974) Figure.2.4.

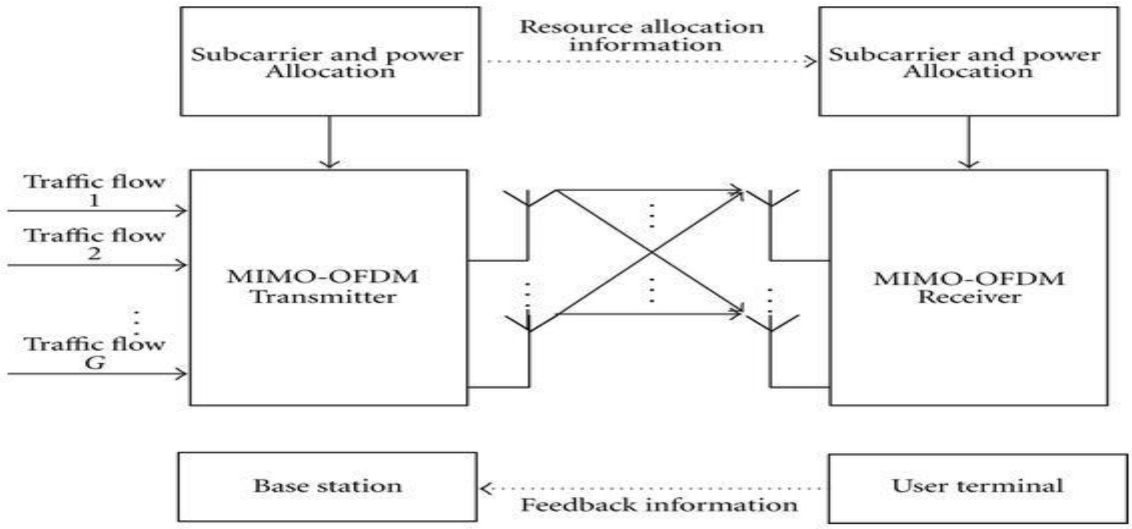


Figure 2.4: MIMO system

2.5.2 MIMO basics

A channel may be affected by fading and this will impact the signal to noise ratio. In turn this will impact the error rate, assuming digital data is being transmitted. The principle of diversity is to provide the receiver with multiple versions of the same signal. If these can be made to be affected in different ways by the signal path, the probability that they will all be affected at the same time is considerably reduced. Accordingly, diversity helps to stabilize a link and improves performance, reducing error rate (Wood et al., 2008). Several different diversity modes are available and provide a number of advantages:

- **Time diversity:** Using time diversity, a message may be transmitted at different times, e.g. using different timeslots and channel coding.
- **Frequency diversity:** This form of diversity uses different frequencies. It may be in the form of using different channels, or technologies such as spread spectrum / OFDM.
- **Space diversity:** Space diversity used in the broadest sense of the definition is used as the basis for MIMO. It uses antennas located in different positions to take advantage of the different radio paths that exist in a typical terrestrial environment.

MIMO is effectively a radio antenna technology as it uses multiple antennas at the transmitter and receiver to enable a variety of signal paths to carry the data, choosing separate paths for each antenna to enable multiple signal paths to be used.

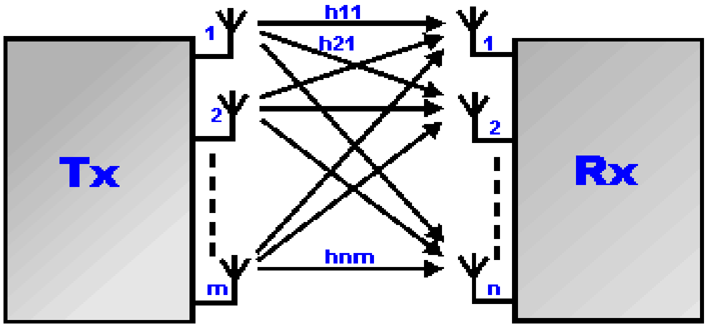


Figure 2.5: General outline of MIMO system

One of the core ideas behind MIMO wireless systems space-time signal processing in which time (the natural dimension of digital communication data) is complemented with the spatial dimension inherent in the use of multiple spatially distributed antennas, i.e. the use of multiple antennas located at different points. Accordingly MIMO wireless systems can be viewed as a logical extension to the smart antennas that have been used for many years to improve wireless (Chang, 1999).

It is found between a transmitter and a receiver the signal can take many paths. Additionally by moving the antennas even a small distance the paths used will change. The variety of paths available occurs as a result of the number of objects that appear to the side or even in the direct path between the transmitter and receiver. Previously these multiple paths only served to introduce interference. By using MIMO, these additional paths can be used to advantage. They can be used to provide additional robustness to the radio link by improving the signal to noise ratio, or by increasing the link data capacity.

The two main configurations for MIMO are given beneath:

- **Spatial differences:** Spatial diversity used in this narrower sense often refers to transmit and receive diversity. These two methodologies are used to provide improvements in the signal to noise ratio and they are characterized by improving the reliability of the system with respect to the various forms of fading.
- **Multiplying the spatial:** This form of MIMO is used to provide additional data capacity by utilizing the different paths to carry additional traffic, i.e. increasing the data throughput capability.

As a result of the use multiple antennas, MIMO wireless technology is able to considerably increase the capacity of a given channel while still obeying Shannon's law. By increasing the number of receive and transmit antennas it is possible to linearly increase the throughput of the channel with every pair of antennas added to the system. This makes MIMO wireless technology one of the most important wireless techniques to be employed in recent years. As spectral bandwidth is becoming an ever more valuable commodity for radio communications systems, techniques are needed to use the available bandwidth more effectively. MIMO wireless technology is one of these techniques.

2.5.3 MIMO formats

There are a number of different MIMO configurations or formats that can be used. These are termed SISO, SIMO, MISO and MIMO. These different MIMO formats offer different advantages and disadvantages - these can be balanced to provide the optimum solution for any given application.

The different MIMO formats - SISO, SIMO, MISO and MIMO require different numbers of antennas as well as having different levels of complexity. Also dependent upon the format, processing may be needed at one end of the link or the other - this can have an impact on any decisions made (Fulford et al., 2002).

2.5.4 SISO, SIMO, MISO, MIMO terminology

The different forms of antenna technology refer to single or multiple inputs and outputs. These are related to the radio link. In this way the input is the transmitter as it transmits into the link or signal path, and the output is the receiver. It is at the output of the wireless link (Fulford et al., 2002).

Therefore the different forms of single / multiple antenna links are defined as below:

- SISO - Single Input Single Output
- SIMO - Single Input Multiple output
- MISO - Multiple Input Single Output
- MIMO - Multiple Input multiple Output

The term MU-MIMO is additionally utilized for a various client form of MIMO as portrayed beneath.

2.5.4.1 MIMO – SISO

The easiest form of radio link can be defined in MIMO terms as SISO - Single Input Single Output. This is effectively a standard radio channel - this transmitter operates with one antenna as does the receiver. There is no diversity and no additional processing required Figure 2.6.

Figure 2.6: SISO - Single Input Single Output

The advantage of a SISO system is its simplicity. SISO requires no processing in terms of the various forms of diversity that may be used. However the SISO channel is limited in its performance. Interference and fading will impact the system more than a MIMO system using some form of diversity, and the channel bandwidth is limited by Shannon's law - the throughput being dependent upon the channel bandwidth and the signal to noise ratio (Engels, 2002).

2.5.4.2 MIMO – SIMO

The SIMO or Single Input Multiple Output version of MIMO occurs where the transmitter has a single antenna and the receiver has multiple antennas. This is also known as receive diversity. It is often used to enable a receiver system that receives signals from a number of independent sources to combat the effects of fading (Wood et al., 2008). It has been used for many years with short wave listening / receiving stations to combat the effects of ionospheric fading and interference Figure 2.7.

Figure 2.7: SIMO - Single Input Multiple Output

SIMO has the advantage that it is relatively easy to implement although it does have some disadvantages in that the processing is required in the receiver. The use of SIMO may be quite acceptable in many applications, but where the receiver is located in a mobile device such as a cellphone handset, the levels of processing may be limited by size, cost and battery drain.

There are two forms of SIMO that can be used:

- **Switched diversity SIMO:** This form of SIMO looks for the strongest signal and switches to that antenna.
- **Maximum ratio combining SIMO:** This form of SIMO takes both signals and sums them to give the combination. In this way, the signals from both antennas contribute to the overall signal.

2.5.4.3 MIMO – MISO

MISO is also called transmit diversity. In this case, the same data is transmitted excessively from the two transmitter antennas. The receiver is then able to receive the optimum signal which it can then use to receive extract the required data Figure 2.8.

Figure 2.8: MISO - Multiple Input Single Output

The advantage of using MISO is that the multiple antennas and the redundancy coding / processing is moved from the receiver to the transmitter. In instances such as cellphone UEs, this can be a significant advantage in terms of space for the antennas and reducing the level of processing required in the receiver for the redundancy coding. This has a positive impact on size, cost and battery life as the lower level of processing requires less battery consumption (DVB, 2009).

2.5.4.4 MIMO

Where there is more than one antenna at either end of the radio link, this is termed MIMO - Multiple Input Multiple Output. MIMO can be used to provide improvements in both channel robustness as well as channel throughput Figure 2.9.

Figure 2.9: MIMO - Multiple Input Multiple Output

In order to be able to benefit from MIMO fully it is necessary to be able to utilize coding on the channels to separate the data from the different paths. This requires processing, but provides additional channel robustness / data throughput capacity.

There are many formats of MIMO that can be used from SISO, through SIMO and MISO to the full MIMO systems. These are all able to provide significant improvements of performance, but generally at the cost of additional processing and the number of antennas used. Balances of performance against costs, size, processing available and the resulting battery life need to be made when choosing the correct option (DVB, 2008)

CHAPTER 3
CLINICAL BACKGROUND

3.1 Medical Importance

Telemedicine is the application of advanced telecommunication technology for diagnostic, monitoring and therapeutic purposes. It enables data transmission from the patient's whereabouts or his/her primary care provider to a specialized medical call center (Reimers, 1996).

Telecardiology is one of the most highly developed of the medical disciplines covered by telemedicine. In addition to the provision of care to patients with heart disease, it has a vital role in educating these patients on the nature of their conditions, improving their compliance to medical therapy, and guiding them in practicing healthy life habits. The benefit of telecardiology in rural communities is especially important because of its capability of overcoming the obstacle of the large distances that would have to be covered in order to access medical assistance Figure 3.1

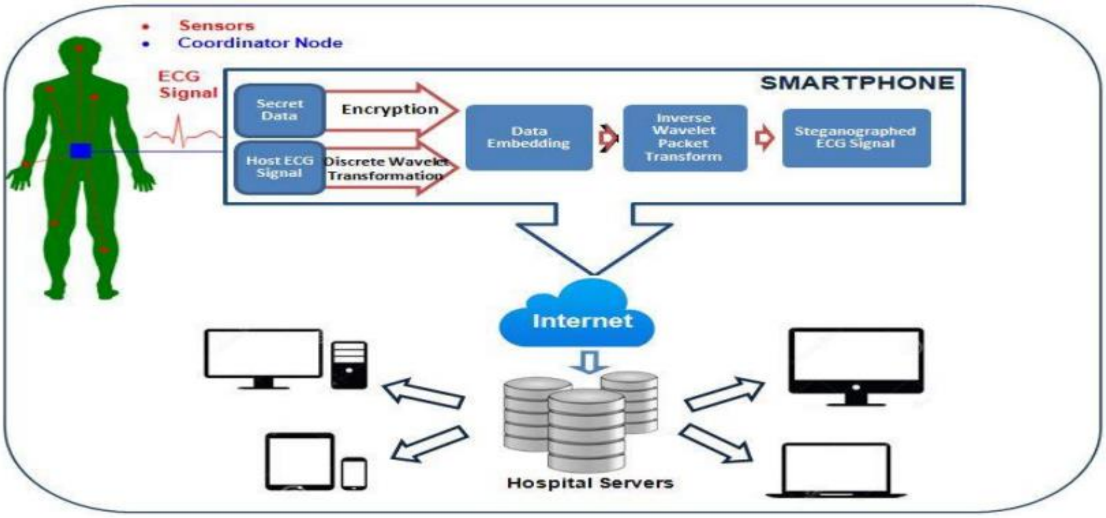


Figure 3.1: Telecardiology

Electrocardiogram (ECG or EKG) is a record of bio-electric potential variation recorded through time on the body surface that represents heart beats [1]. Every heartbeat cycle is normally characterized by the sequence of waveforms known as a P wave, QRS complex and a T wave. Time intervals between those waveforms as well as their shapes and orientation are representing physiological processes occurring in heart and autonomous nervous system. Although today in medical centers advanced equipment and tools are used for detecting heart-beat arrhythmias and other cardiovascular abnormalities, visual inspection of the multi-channel (lead) ECG record is still the first step taken by cardiologists in diagnosis process (Ghaffari et al., 2007) Figure.3.2.

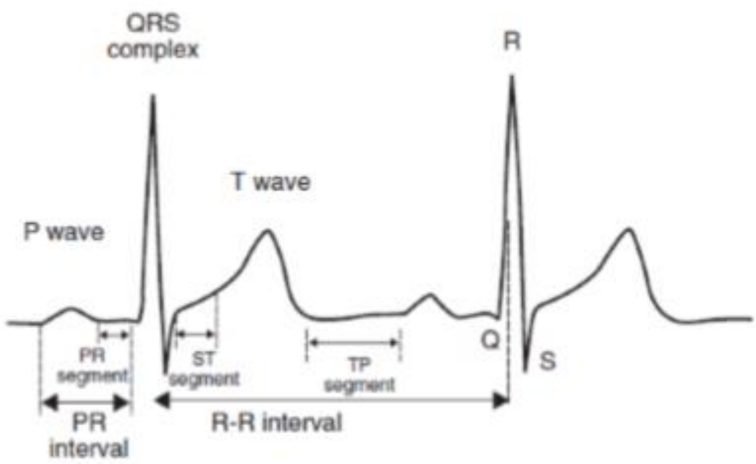


Figure 3.2: Normal ECG signal with marked characters.

Detailed explanation of the physiological process behind the ECG signal shape is out of the scope of this thesis, but for the easier understanding of the main goals of this research a short summary will be given.

Human heart is divided into four main chambers called atria and ventricles both with their left and right instances. Those chambers together form a biological pump for propelling the blood throughout the body. Besides those four obvious sections there are some other parts of the heart for very specialized functions like dividing atria from ventricles, slow impulse propagation, very fast impulse propagation etc., all of them performing particular tasks, ensuring that blood flows properly and efficiently throughout the body. When electrical impulse propagates through heart and all these specialized cells, ECG electrodes pick up that impulse in various directions and speed. In this way ECG waveforms are formed (Clifford et al., 2006) (Camm et al., 2009). With that in mind one can logically assume that different problems in different kind of cells or different parts of heart will have corresponding effects in ECG wave's direction and morphology (Luna, 2008).

Efficient and fast ECG analysis algorithms are needed in clinical practice but also in pre-hospital use cases since clinical findings indicated that there was a significant improvement in patient outcome based on this early treatment (Purvis et al., 1999). Pre-hospital ECG is a test that may potentially influence the management of patients with acute myocardial infarction through wider, faster in-hospital utilization of re-perfusion strategies and greater usage of invasive procedures, factors that may possibly reduce short term mortality (Jakes, 1994). Medical literature suggests clinical importance of ECG not only in identifying heart problems itself, but also other health issues that leave a trace on ECG as a symptomatic phenomenon like ECG patterns reflecting antidepressant treatment.

CHAPTER 4
DIGITAL VIDEO BROADCASTING

The Digital Video Broadcasting (DVB) specifications cover digital services delivered via cable, satellite and terrestrial transmitters, as well as by the internet and mobile communication systems. Digital Video Broadcasting (DVB) is playing a crucial role in digital television and data broadcasting world-wide. DVB services have recently been introduced in Europe, in North- and South America, in Asia, Africa and Australia. Among the more recent achievements are the standards for terrestrial transmission, for microwave distribution and for interactive services via PSTN/ISDN and via (coaxial) cable (Reimers, 1996). As it is stated by the standard in (DVB, 2008) techniques used by DVB are able to deliver data at approximately 38 Mbit/s within one satellite or cable channel or at 24 Mbit/s within one terrestrial channel. The satellite member of the DVB family, DVB-S, is defined in European Standard EN 300 421 (ETSI, 1997)). September 1993, and at the end of the same year produced its first specification, DVB-S (DVB, 2009), the satellite delivery specification now used by most satellite broadcasters around the world for DTH (direct-to-home) television services. The DVB-S system is based on QPSK modulation and convolutional forward error correction (FEC), concatenated with Reed-Solomon coding. In 1998, DVB produced its second standard for satellite applications, DVB-DSNG (DVB, 2010), extending the functionalities of DVB-S to include higher order modulations (8PSK and 16QAM) for DSNG and other TV contribution applications by satellite.

4.1 DVB Family

DVB contains a suite of standards for digital television. All of these standards are maintained by DVB Project. DVB Project is formatted from a union of European broadcasters, equipment manufacturers and other regulatory bodies in September 1993 and the purpose of this union was to agree in which technology will be used in digital broadcasting. Currently DVB project standards are used in more than 35 countries (mostly Europe) (DVB, 2009). DVB standard are “divided” in three “traditional” standards which are DVB-S for satellite systems as in Figure 4.1, DVB-C for link systems and lastly DVB-T for terrestrial networks. Except for these main standards, DVB project have some supporting standards which are needed to cover areas such as service information (DVB-SI) and subtitles (DVB-SUB). In recent years, DVB added standards for new technological areas such as handheld devices (DVB-H) and mobile TV

(DVB-SH) making the DVB family standards even bigger. At the end of 2009 DVB upgraded two of first generation broadcasting standards to second generation standards (DVB-S2 and DVB-T2). Currently DVB's target is to upgrade the last of the first generation transmission standards (DVB-H).

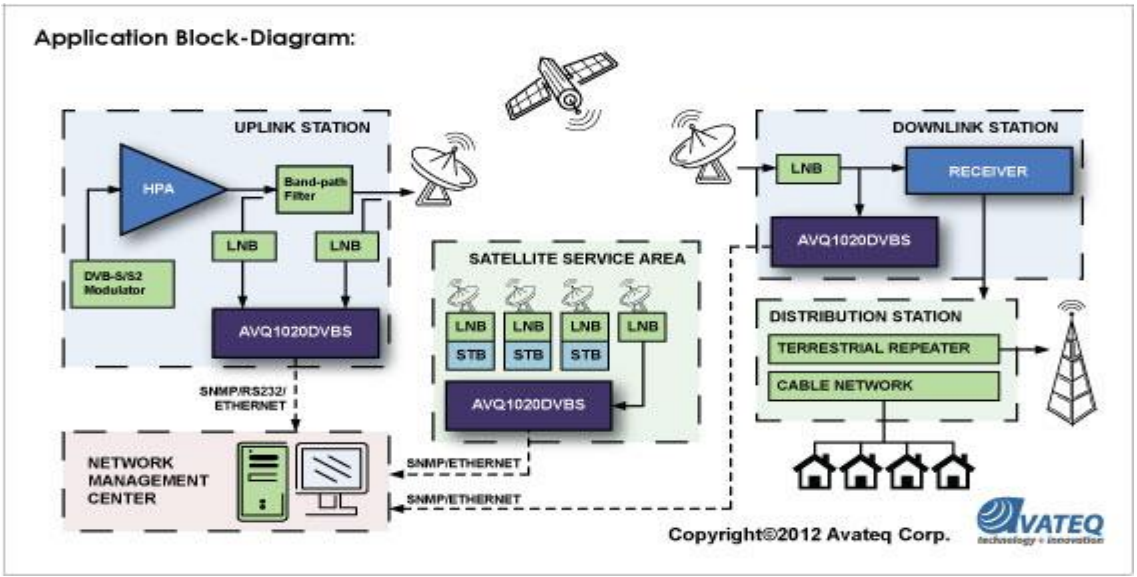


Figure 4.1: DVB-S system

4.1.1 DVB – T

DVB-T (Digital Video Broadcast – Terrestrial) is the most common used set of standards in the world (mostly in Europe) for the terrestrial TV transmission. It was first published in late 1997 and now is used in more than 35 countries. DVB-T is designed to be compatible with MPEG-2 coded TV signals (and lately MPEG-4 but not all of MPEG-4 standards) as well as audio encoding systems and to work within the existing VHF and UHF The transmission is based on Coded Orthogonal Frequency Division Multiplex (COFDM). COFDM uses a large number of carriers. Each of these carriers is used to transmit only a portion of the total amount of data. The data is modulated on the carriers with QPSK or QAM. COFDM has the advantage that it is very robust against multipath reception and frequency selective fading. This robustness against multipath reception is obtained through the use of a 'guard interval'. This is a proportion of the time there is no data transmitted. This guard interval reduces the transmission capacity (Shannon et al., 1990).

Because of this multipath immunity, it is possible to extend the coverage area with the use of an overlapping network of transmitter stations which use the same frequency, a so-called single frequency network (SFN). In the areas of overlap, the weaker of the two signals is considered as an echo due to multipath reception. However, the stations have to be synchronized and the echo has to fall within the guard time. Hence, if two stations are far apart, the time delay between the two signals can be large and the system will need a large guard interval (Chang, 1970).

There are two COFDM transmission modes possible in the DVB-T system. A 2k mode which uses 1705 carriers and a 8k mode which uses 6817 carriers. The 2k mode is suitable for single transmitter operation and for relatively small single frequency networks with limited transmission power. The 8k mode can be used both for single transmitter operation and for large area single frequency networks. The guard interval is selectable.

Portable and mobile reception of DVB-T signals is possible. It is even possible to mix the reception modes by using hierarchical transmissions, in which one of the modulated streams (so-called HP – High Priority stream), is given a higher protection against errors, to make is suitable for mobile reception; while the other one (so-called LP – Low Priority stream), has a lower protection. The higher protection mode will have a lower net bit rate available.

4.2 DVB – T coding and modulation

In the transmitter Figure 4.1, the first step is the encoding of image, audio and other data (if those exist) and then multiplexing it with the MPEG-2 PS. One or more MPEG-PSs are joined together into an MPEG transport stream (MPEG-TS).

The transfer rate starts from 5Mbit/sec and could rise to the 32Mbits/sec depending on configuration and coding we choose in relation to the application we want to use.

Therefore, the selection of the video quality is the end users choice. First, the bit stream is divided into sections of 188 bytes. To these we apply Reed-Solomon RS coding which provides error correction up to 8 bytes (for each of these packages). These packages are then modulated using QPSK (Quadrature Phase Shift Keying), 16QAM (Quadrature Amplitude Modulation) or 64QAM While higher order modulation rates are able to offer much faster data rates and higher levels of spectral efficiency for the radio communications system, this comes at a price. The higher order modulation schemes are considerably less resilient to noise and interference (DVB, 2009). QPSK is a phase modulation algorithm. It's an improved algorithm based on RSK configuration, using 4 possible states: 45, 135 225 and 315 degrees. QAM configuration combines two signals into one channel. The final version of the QAM signal will have two carriers on the same frequency that differ in phase by 90 degrees. After the modulation the symbols are grouped in the blocks. Each of these blocks can have 1512, 3024 or 6048 symbols. 68 of these blocks are called one frame. After the blocks are formed by the algorithm OFDM (Orthogonal Frequency Division Multiplexing) or COFDM (coded OFDM) guard intervals of length 1/4, 1/8, 1/16 or 1/32 of total length of the block are imported. After that step the final signal is modulated from digital to analog and transmitted in the air medium using a pre-set bandwidth 6MHz, 7MHz or 8MHz. The reason we use the guard intervals is to avoid the ISI (inter-symbol Interference) phenomenon that occurs when a signal reaches the receiver using two different paths and resulting in weakening the signal. By importing the guard intervals in our signal in a default time rate we succeed to synchronize our receiver with our signal. This leads to a minor drawback, the small increase in the non-useful information we receive (DVB, 2008).

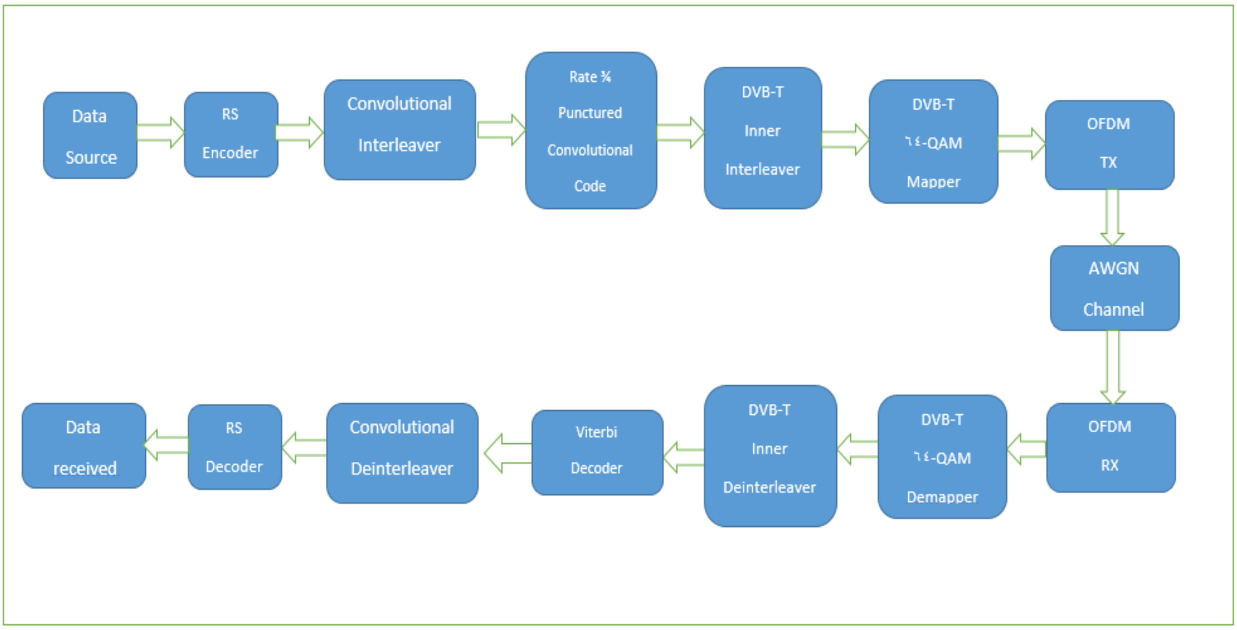


Figure 4.2: Block diagram of DVB-T system

4.3 Second Generation Digital Video Broadcasting over Satellite (DVB-S2)

Digital satellite transmission technology has evolved considerably since the publication of the original DVB-S specification. New coding and modulation schemes permit greater flexibility and more efficient use of capacity, and additional data formats can now be handled without significant increase of system complexity. DVB-S2 Figure 4.3 has a range of constellations on offer. DVB-S2 supports a wide range of modulation schemes, including QPSK (2bits/symbol), 8PSK (3bits/symbol), 16APSK (4bits/symbol) and 32APSK (5bits/symbol). These APSK modulation schemes provide superior compensation for transponder non-linearity's than QAM. DVB-S2 is so flexible that it can cope with any existing satellite transponder characteristics, with a large variety of spectrum efficiencies and associated SNR requirements. Furthermore it is designed to handle a variety of advanced audio video formats which the DVB Project is currently defining (DVB, 2008).

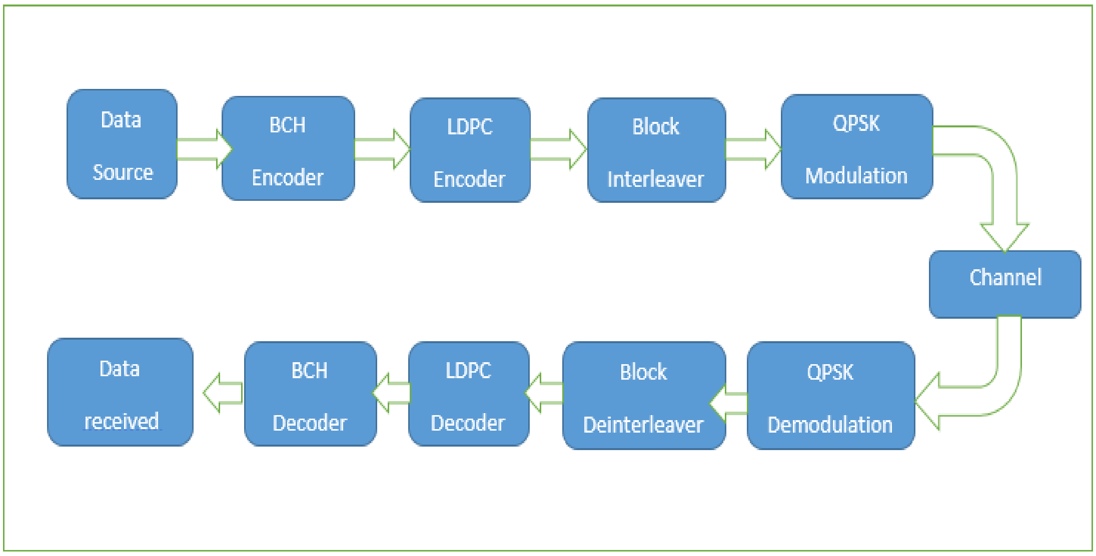


Figure 4.3: Block diagram of DVB-S2 system

Next, we will explain each block of Figure 4.3.

4.3.1 Data source

This the block represent the data of the ECG signal after being transformed into digital.

4.3.2 BCH encoder/decoder

One of DVB-S2 standard advances is the forward error correction which is deployed to reduce BER in transmissions is BCH error correction. Output of BBFrame buffering block at the sender side, as above mentioned, are frames of K_{bch} bits where a BCH (N_{bch}, K_{bch}) error correction with the correcting power of t will be applied to them. For each of 11 rate of coding presented in the standard K_{bch} and N_{bch} values are defined including the t-error correcting parameter. In Table 4.1 these values are shown for normal and short frames respectively.

The output of BCH encoder called BCHFEC frame will be created by adding parity check bits to make a frame with N_{bch} size. N_{bch} is the input of inner LDPC encoder which is also named K_{ldpc}.

4.3.3 LDPC encoder/decoder

N_{bch}, the BCH encoder output as the input of inner FEC encoder will be processed at LDPC encoder to be protected from error with parity bits. The number of parity bits are given in Tables 4.1 as:

$$\text{Number of LDPC parity bits} = N_{ldpc} - N_{bch}.$$

LDPC encoder supports 11 coding rates. These coding rates are the ratio between information bits (N_{bch} bits) and LDPC coded block bits which is the FECFRAME.

Table 4.1: Coding parameters for normal FECFRAME N_{ldpc}=64800

LDPC Code	BCH Uncoded Block K_{bch}	BCH coded block N_{bch} LDPC Uncoded Block k K_{ldpc}	BCH t-error correction	LDPC Coded Block N N_{ldpc}
1/4	16008	16200	12	64800
1/3	21408	21600	12	64800
2/5	25728	25920	12	64800
½	32208	32400	12	64800
3/5	38688	38880	12	64800
2/3	43040	43200	10	64800
3/4	484080	48600	12	64800
4/5	51648	51840	12	64800
5/6	53840	54000	12	64800
8/9	57472	57600	12	64800
9/10	58192	58320	12	64800

For example for rate 1/4 in a normal frame it means that for every 1 bit of information sent from outer FEC coder (BCH), there will be 3 bits of parity checks added in LDPC encoder. The lower this ratio the more protection of data against error has been carried out in LDPC encoder. This will result in more robust data transmission, and it will reduce system throughput indeed. At the receiver side, LDPC decoder will check the received sequence till the parity checks are satisfied up to 50 iterations. This error correction uses the sparse parity-check matrices with a hard decision making algorithm.

4.3.4 Block interleaving/deinterleaving

Interleaving process is the next step in DVB-S2 for modulations 8PSK, 16APSK and 32APSK. Interleaving on QPSK is not going to be done and as for DVB-S2 model, 16APSK and 32APSK modulations are not included so we will discuss them in our proposed model later.

Interleaver block in will make this an 8PSK by writing column wise serially the output of LDPC encoder in a 3 by n ldpc (21600 for 8PSK) matrix and then will read it out row wise. The MSB of BBHeader will be read-out first since for rate 3/5 it will be read-out as third.

Interleaving process creates rows in a matric from the LDPC encoder output according to the modulation order M, so each row will contain a symbol ready to be mapped in the next block, modulation. At the receiver side de-interleaver block will receive the output of demodulator block as input and will apply the reverse process to create a serial output for the LDPC decoder input.

4.3.5 QPSK modulation/demodulation

Modulation block will process the interleaved vector by first mapping each row to a symbol which in our case is a gray mapping, then the mapped symbols will be assigned to constellations.

4.3.6 Channel

The channel is the medium between the transmitter and receiver and it can be different types. One of the most famous channel is the AWGN which represent the simple channel as the signal suffer from only the additive white Gaussian noise (Reimers, 1990).

Another type is the MIMO channel which is represent multiple channel with different condition.

CHAPTER 5
SIMULATION AND PERFORMANCE ANALYSIS

5.1 Introduction

In this chapter, two Telecardiology systems are assumed and simulations are run to see the effects of transmitting ECG signals through these systems. Specifically, transmissions through DVB-T and DVB-S2 systems are analyzed. MIMO is then added to the DVB-T system to see how it changes the performance.

5.2 ECG signal generation

An ECG signal is obtained with the assistance of ECG electrodes. ECG sensors can be placed at the midsection of the patient or generally at the wrist.

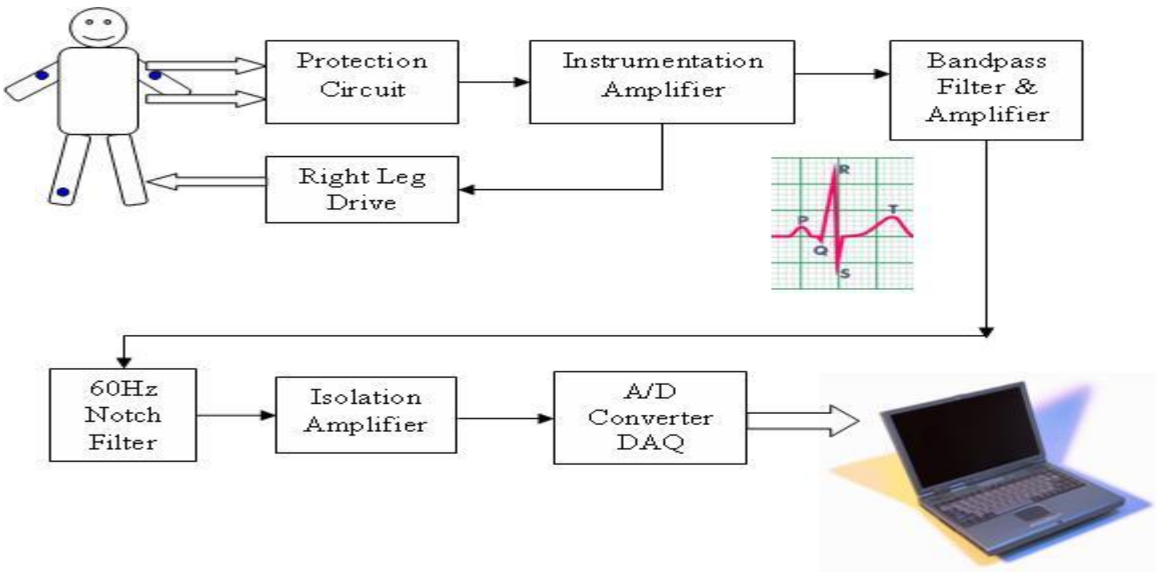


Figure 5.1: ECG system

This thesis does not use real ECG data. Therefore, first step in Telecardiology transmission simulation becomes the ECG signal generation. In this section, we will show how computer generated ECG signals for two different patients with different heart rates Figure 5.1 and Figure 5.2 can be transformed from analog into digital.

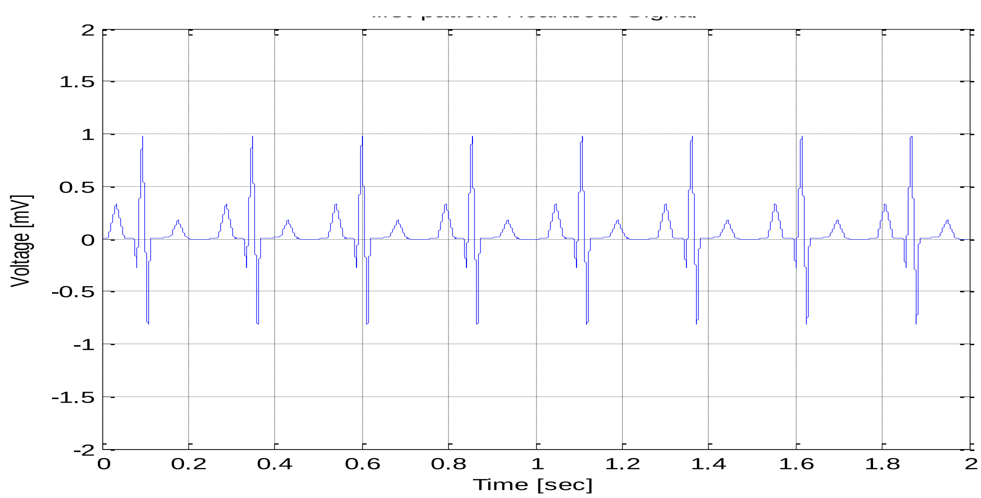


Figure 5.2: ECG signal of the first patient

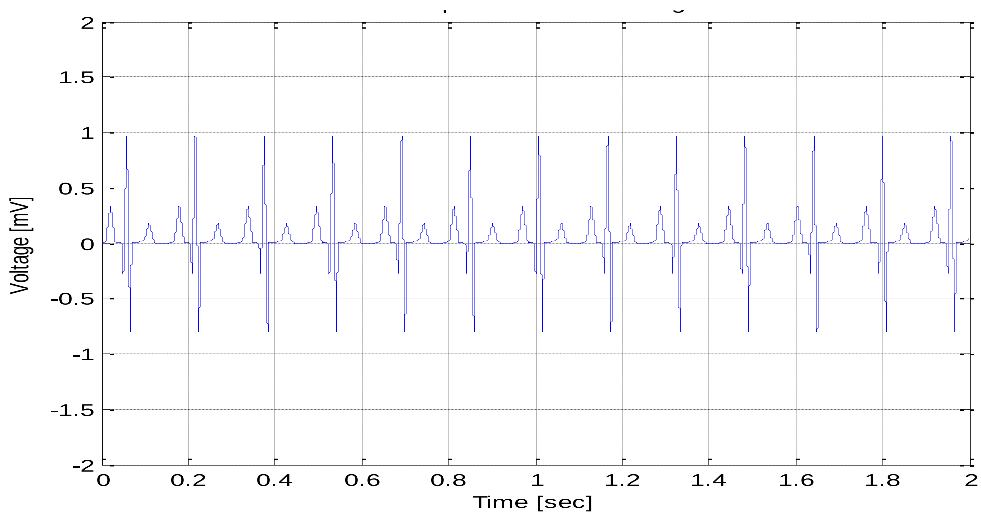


Figure 5.3: ECG signal of the second patient

The procured ECG signals from the sensors are in analog form. These analog signals are sent to the analog to digital converter (ADC) unit for conversion into computerized signal. In this way, analog signals are changed over into digital signals. Conversion of analog signal into digital includes the following three steps:

1. Sample the signal with high rates to capture all the changes in the ECG signal.
2. Quantize the sampled signal to approximate each sample to its proper level.
3. Encode the signal into binary data.

Figure 5.4 shows the sampled ECG signal for the first patient and it is obtained by using Simulink program. Sampling rate is taken to be 3 kHz as the input signal is not more than 200 Hz as the maximum heart rate for normal people when do running. According to Nyquist equation the sampling frequency should be greater than double the input signal maximum frequency. This allows us catch any variation on the signal like the unusual ECG signal for people with heart problems.

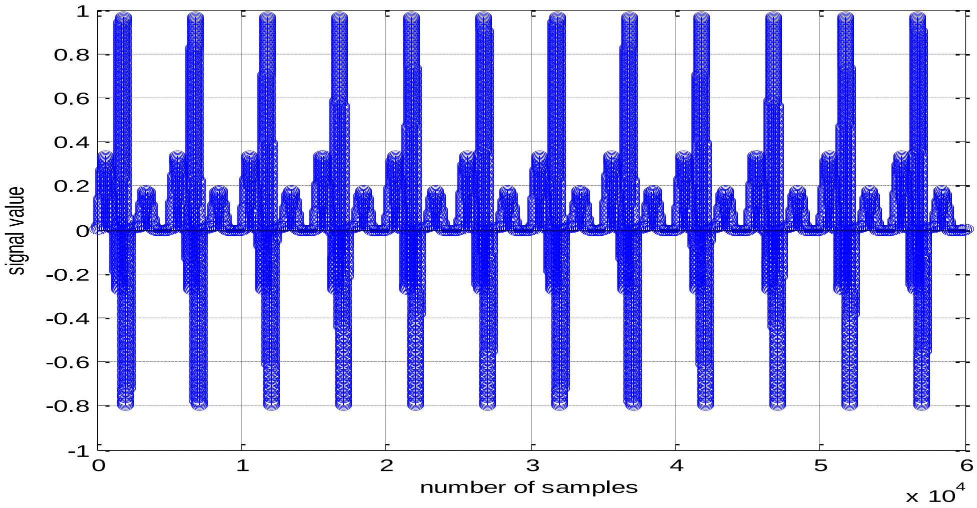


Figure 5.4: Sampled representation of ECG signal

Second step in conversion is quantization. Figure 5.5 shows the ECG signal after being quantized.

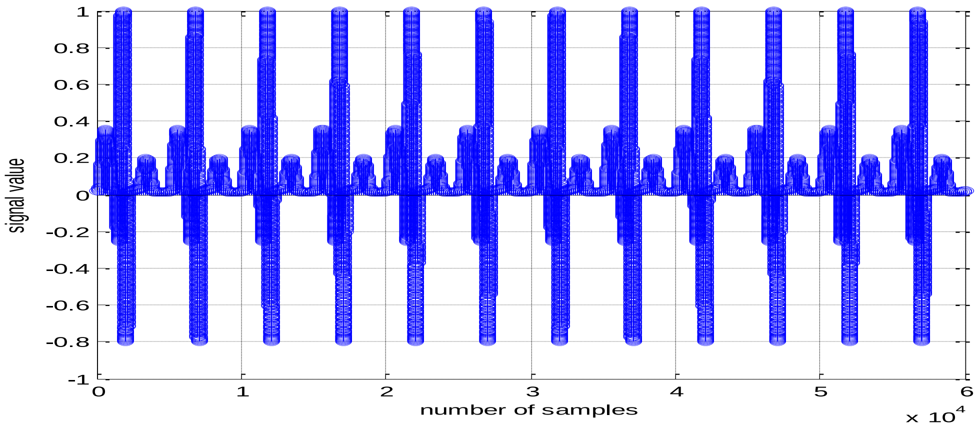


Figure 5.5: Quantized ECG signal

Figure 5.6 shows the difference (error) due to quantization between the two signals. Maximum error is almost 0.032.

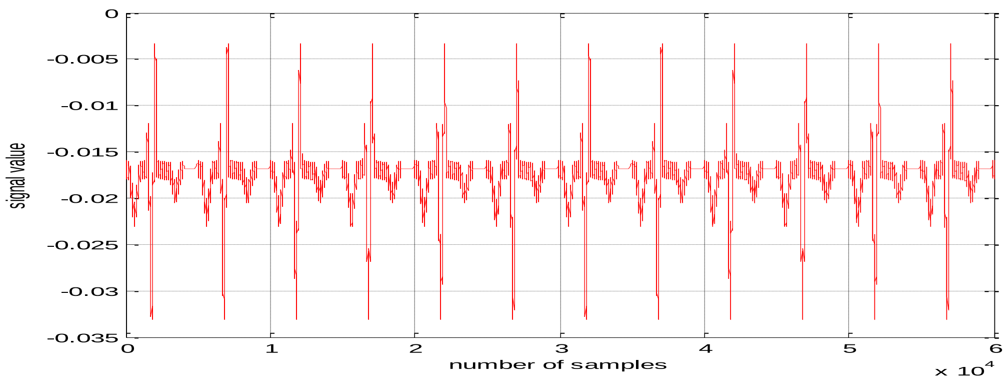


Figure 5.6: Error between quantized and sampled ECG signal

Figure 5.7 represents the ECG signal after being encoded into 8 bit codeword.

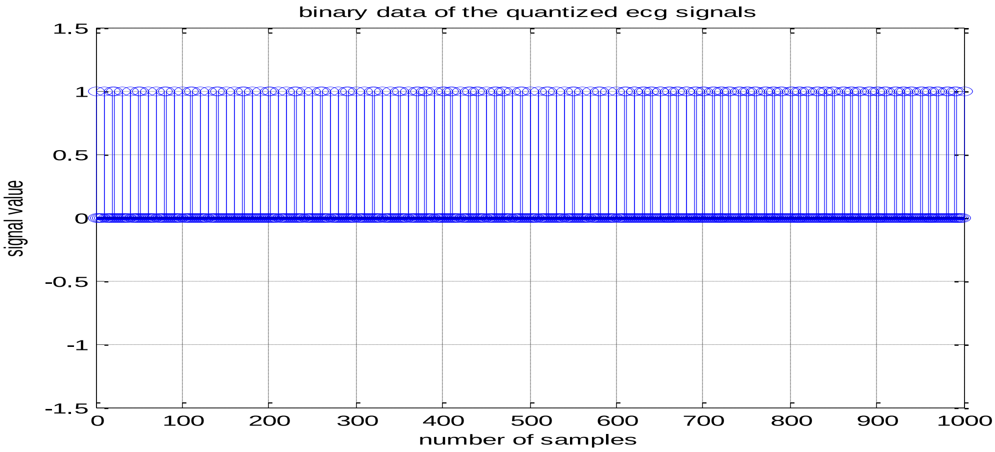


Figure 5.7: Binary representation of ECG Signal

5.3 Comparison of DVB-T with 64-QAM in AWGN Channel

One aim of this section is to analyze the performance of ECG transmission of Telecardiology system using DVB-T technology with using 64-QAM modulation scheme in an additive white Gaussian noise (AWGN) channel. The result is compared to theoretical bit error rate (BER) of 64-QAM, the modulation type chosen in DVB-T, in an AWGN channel. Figure 5.8 shows this comparison.

BER Bit Error Rate Tutorial and Definition

Bit error rate, BER is used to quantify a channel carrying data by counting the rate of errors in a data string. It is used in telecommunications, networks and radio systems.

$$\mathrm{BER} = \frac{Number\ of\ errors}{Total\ number\ of\ bits}$$

.

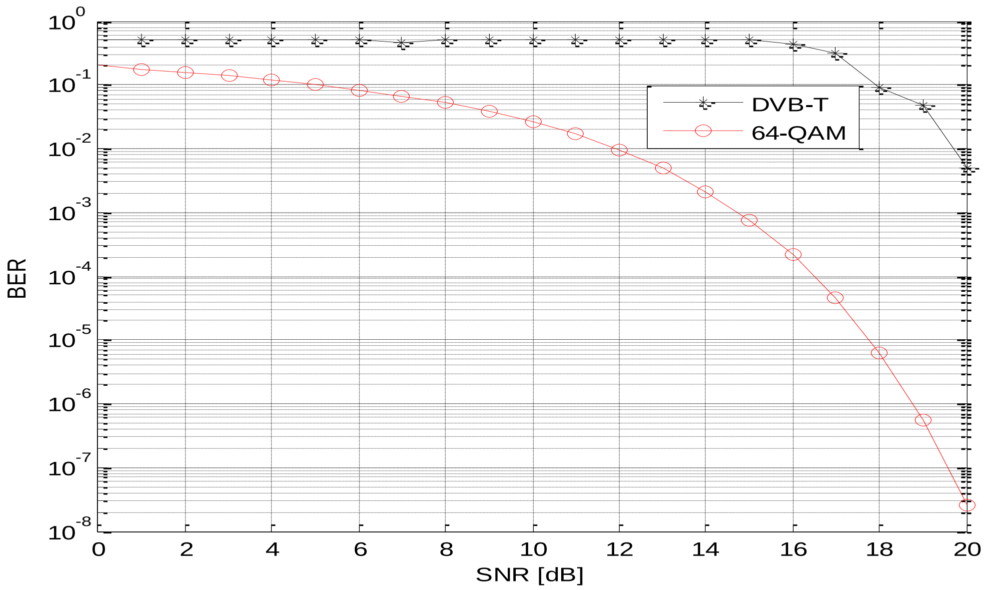


Figure 5.8: Comparison between DVB-T and theoretical 64-QAM in an AWGN channel

It can be seen from Figure 5.8 that DVB-T technology is 7 dB worse at 10^{-2} BER when compared to the performance of theoretical 64-QAM.

5.4 Comparison between DVB-S2 and QPSK in AWGN Channel

In this section, a comparison is conducted between the performance of ECG transmission using DVB-S2 technology with QPSK modulation scheme and theoretical BER of QPSK in an AWGN channel see Figure 5.9. QPSK is the modulation type chosen in DVB-S2.

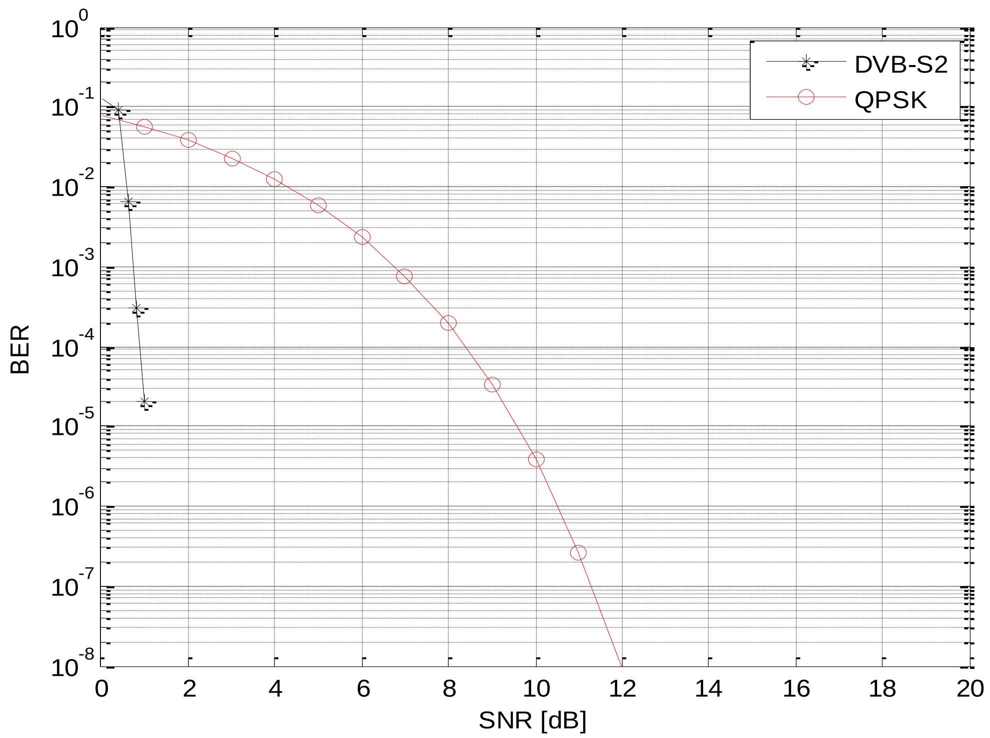


Figure 5.9: Comparison between DVB-S2 and theoretical QPSK in an AWGN channel

Figure 5.9 shows that DVB-S2 technology is about 8.5 dB better than theoretical QPSK at a BER of 10^{-5}.

5.5 Comparison between DVB-T and DVB-S2 in AWGN Channel

One of the goals of this thesis is to see if a terrestrial or a satellite technology is more suitable for Telecardiology. One way to check this is to observe the system performance during the transmission of ECG signals using DVB-S2, a satellite technology, and DVB-T, a terrestrial technology, in an AWGN channel. Figure 5.10 shows the performance comparison of ECG transmission using DVB-S2 and DVB-T technologies in an AWGN channel.

It is shown that DVB-S2 performs much better than DVB-T in an AWGN channel. The improvement is about 18-19 dB at 10^{-2} BER. The reasons for this most probably are the usages of better error coding and modulation methods in DVB-S2.

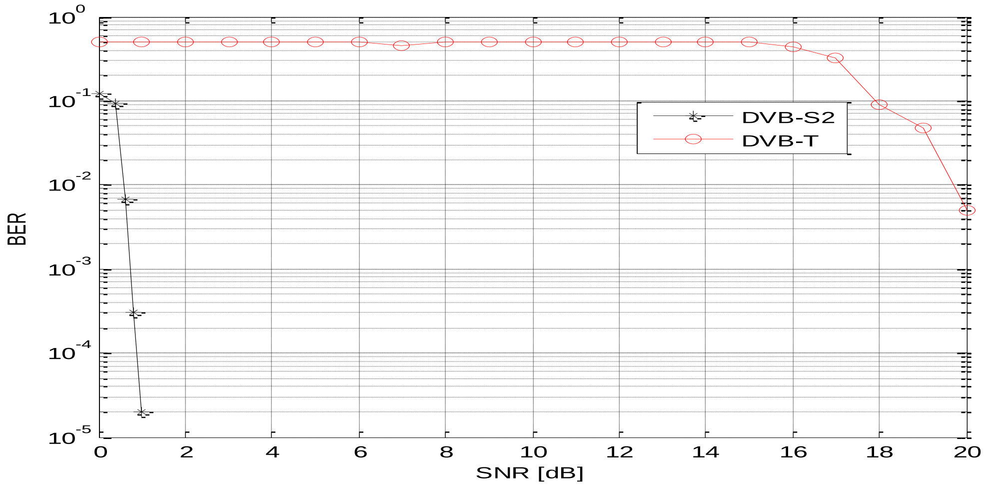


Figure 5.10: Performance of DVB-S2 and DVB-T in AWGN channel

5.6 Effect of MIMO on DVB-T Performance

In order to analyze how much of an improvement is obtained with the use of MIMO, performance of transmission of ECG using DVB-T technology is compared to the performance when MIMO is additionally included. This is shown in Figure 5.11, which clearly indicates an about 5 dB gain of MIMO inclusion at 10^{-2} BER.

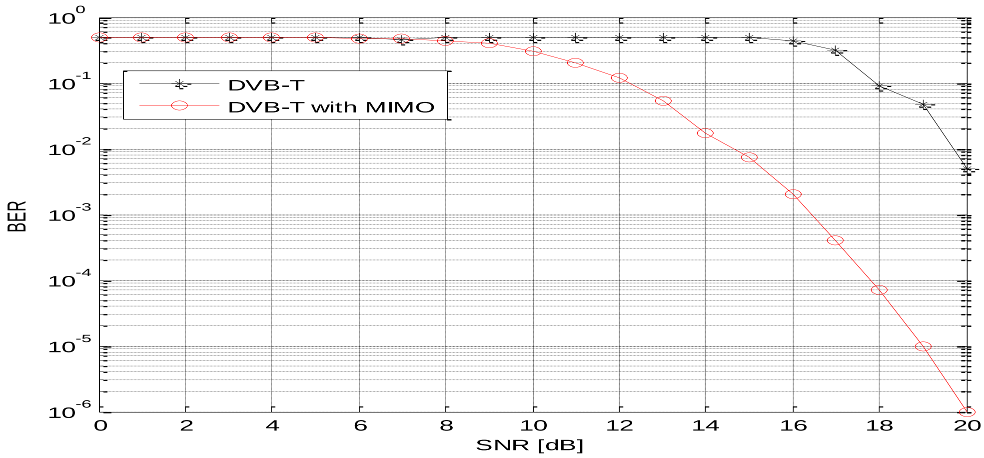


Figure 5.11: Effect of MIMO on DVB-T performance

CHAPTER 6
CONCLUSION

In this thesis we analyze if terrestrial or satellite transmission is better suited for ECG transmission for remote heart monitoring in Telecardiology. It is hoped that using this technology remote medical professionals such as doctors can accurately read and interpret the ECG of a patient.

Specifically, DVB-T and DVB-S2 technologies are used for ECG transmission, after being converted into digital. Simulations are carried out over Additive White Gaussian Channel (AWGN) and bit error rate (BER) results are analyzed. Results are compared to theoretical values as well. It is observed that for remote ECG transmission, performance improves by 18 dB if DVB-S2 technology is used over DVB-T technology.

It is known that wireless transmission suffers from multipath propagation. This is due to signals reaching the receiver through different paths. One of the techniques for reducing the effects is Multiple Input Multiple Output (MIMO) antenna technology. This thesis analyzes the performance of ECG transmission through a multipath Rayleigh fading channel and shows that with the addition of MIMO a performance gain of 5 dB is obtained.

Overall, DVB-S2 technology is shown to be superior to DVB-T for ECG transmission, both in AWGN and in fading channels. However, it is known that satellite transmission suffers from more delay and degradations than terrestrial transmission. Therefore, a trade-off exists such that if performance loss can be tolerated then DVB-T technology might be used instead for faster ECG transmission.

Future work will include analyzing other satellite, cable or terrestrial DVB standards for use with ECG transmission as well as use of other techniques for mobile wireless systems, and also applying equalization, and multipath fading reduction algorithms.

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APPENDIX
SOURCE CODE

Source codes for the simulation models used in this thesis are given below:

A.1. Create heart beat data

This code is used to generate the heart beat data.

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
clc
clear all
close all
x1 = 1*ecg(8000).';
y1 = sgolayfilt(kron(ones(1,26),x1),0,21);
n = 1:63168;
del = round(50*rand(1));
mhb = y1(n + del);
t = 0:3.1662e-5:2;
data.time=t';
data.signals.values =mhb;
data.signals.dimensions =1;
save('ecg','data');
save('hany','t','mhb')
plot(t,mhb);
axis([0 2.5 -2 2]);
grid;
xlabel('Time [sec]');
ylabel('Voltage [mV]');
title('Maternal Heartbeat Signal');
```

A.2. Create digital data for Simulink

This MATLAB code is used to convert the ECG signal from analog into digital.

%%%

```
clc
clear all
close all
load heart_beat
x=mhb;%the samples
N=10;% no of bit for each sample
R=2^N;% the no of qunatized level
a=linspace(-1,1,R); % rangens for N bit
b=0:R-1;
y=quantiz(x,a,b);
xq=(y/max(y)-.5)*2;
e=x-(xq);
y1=de2bi(y);
y2=reshape(y1',1,numel(y1));
S1=1504;
T1=211;
y3=y2(1:S1*T1);
y4=reshape(y3,S1,T1);
N1=zeros(S1,1,T1);
for ii=1:T1
    N1(:,:,ii)=y4(:,ii);
end
sample=logical(N1);
step=0.0476;
```

```
tout=0:step:10;
y3=timeseries(sample,0:step:10);
% save the varaiable for the simulink
save('w_ecg','tout','sample')
save('w_ecg1','y3')
plot(x,'b')
title('the ecg signal')
xlabel('the samples')
ylabel('signal value')
grid on
figure
plot(xq,'b')
title('the quantized ecg signal')
xlabel('the samples')
ylabel('signal value')
grid on
figure
plot(e,'r')
grid on
title('the error between quantized and ecg signals')
xlabel('the samples')
ylabel('signal value')
xlabel('the samples')
ylabel('signal value')
figure
stem(y2(1:1000))
grid on
title('binary data of the quantized ecg signals')
xlabel('the samples')
ylabel('signal value')
```

```
axis([0 1000 -1.5 1.5])
```

A.3.Create comparison figures

This MATLAB code creates the comparison figures

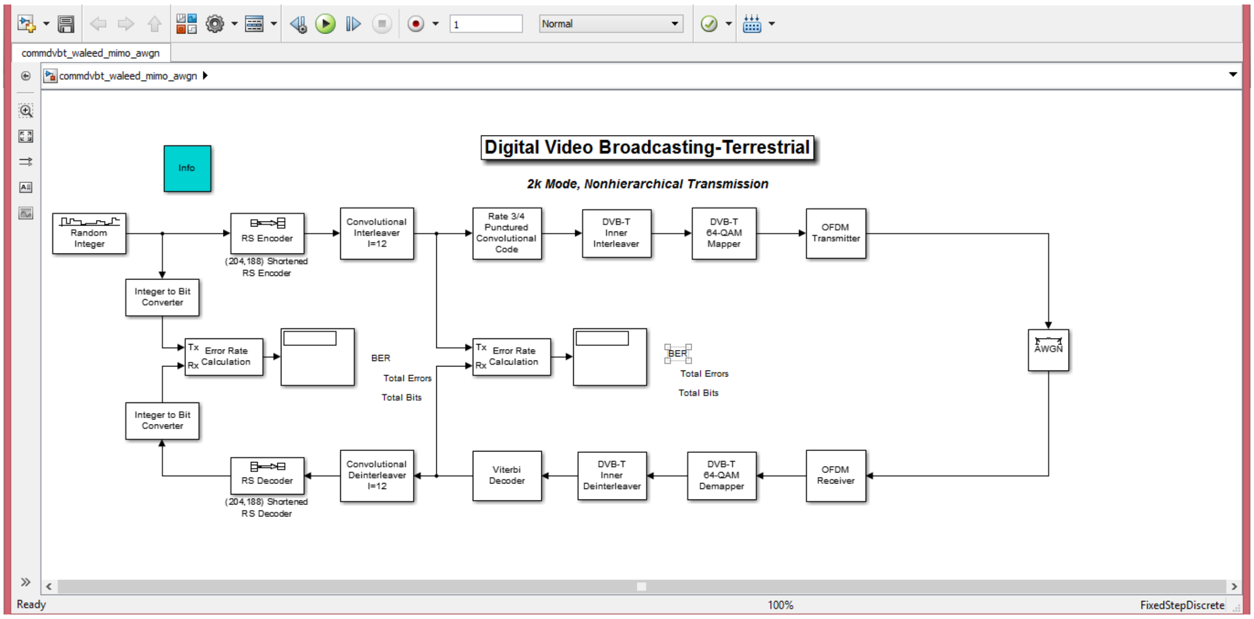
```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
clc
clear all
close all
%%%%%% the DVB S2 with our ECG data
load('default qpsk_64qam.mat')
snr=0.1:0.1:1;
ber_s2=[0.1238 0.1176 0.1107 0.1028 0.09281 0.08156 0.05252
0.006651 0.0003 0.00002];
ber_s2_m=1e-7*ones(1,10);
ebn0=0:20;
snr1=[snr 2 3 4 5 6 7 8 9 10 12 14 16 18 20];
ber_t_m=[0.493 0.4916 0.4909 0.4901 0.4893 0.4886 0.4877 0.486
0.485 0.482 0.471 0.444 0.396 0.3 0.2061 0.1176 0.05362 0.0174
0.0075  0.002 0.0004 0.00007  0.00001 0.000001];
ber_t_m1=[0.493 0.4893 0.4886 0.4877 0.486 0.485 0.482 0.471
0.444 0.396 0.3 0.2061 0.1176 0.05362 0.0174 0.0075  0.002
0.0004 0.00007  0.00001 0.000001];
ber_t=[0.499 0.499 0.499 0.499  0.499 0.499  0.499 0.499 0.499
0.499 0.449 0.498 0.498 0.498 0.498 0.498 0.497 0.497 0.497
0.44 0.3205 0.090 0.048 0.005];
ber_t1=[0.499 0.499 0.499 0.499 0.499 0.499 0.499 0.449 0.498
0.498 0.498 0.498 0.498 0.497 0.497 0.497 0.44 0.3205 0.090
0.048 0.005];
snr2=[0 1 2:20];
%%%%%%%%%%%%%%%%
figure
semilogy(snr2,ber_t1,'k*-')
hold on
semilogy(ebn0,ber_qam,'ro-')
```

```
grid on
xlabel('SNR')
ylabel('BER')
legend('the DVB-T ','64-QAM')
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
figure
semilogy(snr,ber_s2,'k*-')
hold on
semilogy(ebn0,ber_qpsk,'ro-')
grid on
xlabel('SNR')
ylabel('BER')
legend('DVB-S2 ','QPSK')
axis([0 20 1e-8 1])
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
figure
semilogy(snr2,ber_t_m1,'r*-')
grid on
xlabel('SNR')
ylabel('BER')
%title('the DVB t-mimo result')
legend('DVB-T with MIMO')

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

figure

semilogy(snr,ber_s2,'k*-')
hold on
semilogy(snr2,ber_t1,'ro-')
```

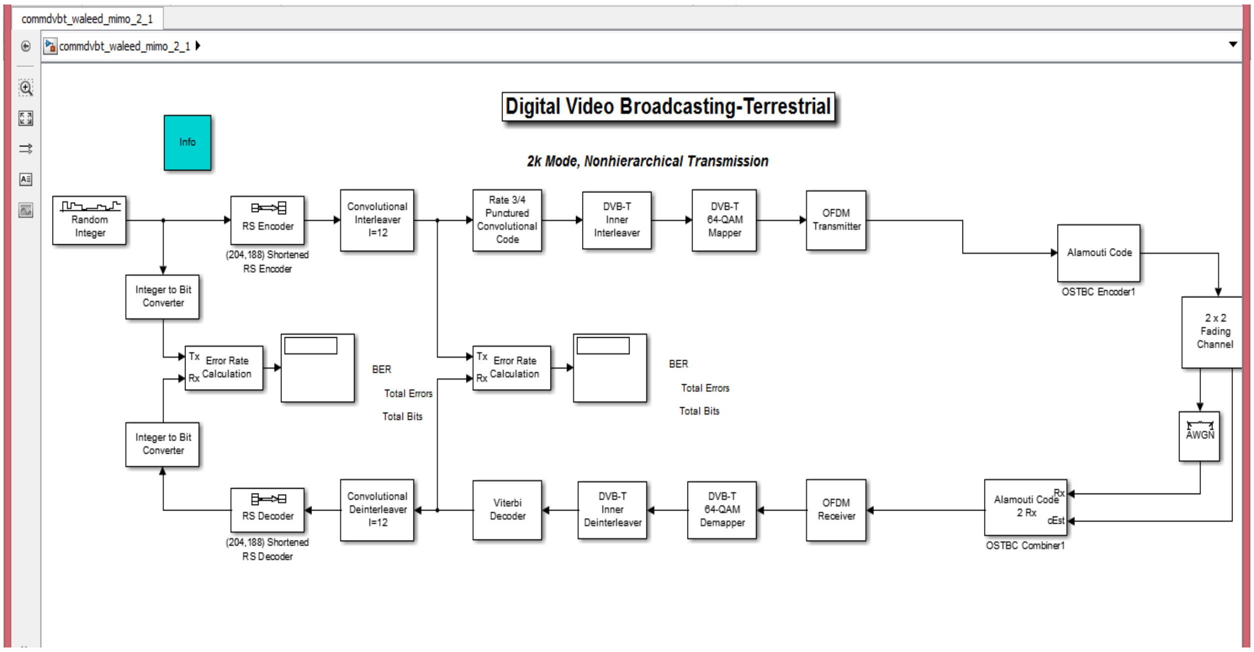
```
grid on
xlabel('SNR')
ylabel('BER')
%title('the DVB comparisson result')
legend('DVB-S2 ','DVB-T')

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
figure
semilogy(snr2,ber_t1,'k*-')
hold on
semilogy(snr2,ber_t_m1,'ro-')
grid on
xlabel('SNR')
ylabel('BER')
%title('the DVB t result')
legend('DVB-T ','DVB-T with MIMO')

%%%%%%%%%%%%%%%%%%%%%%%%%%
figure
semilogy(snr,ber_s2,'k*-')
hold on
semilogy(snr2,ber_t_m1,'ro-')
grid on
xlabel('SNR')
ylabel('BER')
%title('the DVB t result')
legend('DVB-S2 ','DVB-T with MIMO')
```

%%

A.4. Simulink model for DVB-T with AWGN



A.5. Simulink model for DVB-T with MIMO:



A.6. Simulink model for DVB-S2 with AWGN