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PATTERN BASED COMMUNICATION: AN INTEGRATED MODULATION AND CODING TECHNIQUE DESIGNED FOR ADAPTIVE COMMUNICATION IN ADDITIVE WHITE GAUSSIAN NOISE CHANNEL

Ph.D. THESIS

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ÖRÜNTÜ TABANLI İLETİŞİM: TOPLANIR BEYAZ GAUSS GÜRÜLTÜ KANALINDA UYARLANABİLİR HABERLEŞME İÇİN TASARLANMIŞ TÜMLEŞİK KİPLEME VE KODLAMA TEKNİĞİ

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To my spouse and my son,

FOREWORD

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ABBREVIATIONS

A	: Amplitude
ADC	: Analogue-to-Digital Converter
AFPSK	: Amplitude Frequency Phase Shift Keying
AM	: Amplitude Modulation
AMC	: Adaptive Modulation and Coding
AMPS	: Advanced Mobile Phone System
ANN	: Artificial Neural Network
ASK	: Amplitude Shift Keying
ATU	: Alliance of Telecommunication Industrial Solution
AWGN	: Additive White Gaussian Noise
BER	: Bit Error Rate
bps	: bits per second
BPSK	: Binary Phase Shift Keying
CDMA	: Code Division Multiple Access
CN	: Cognitive Network
CP	: Cyclic Prefix
CPFSK	: Continues Phase Frequency Shift Keying
CPM	: Continues Phase Modulation
CR	: Cognitive Radio
DAC	: Digital-to-Analogue Converter
DFT	: Discrete Fourier Transform
DS	: Direct Sequence
DSSS	: Direct Sequence Spread Spectrum
DSP	: Digital Signal Processor
F	: Frequency
FCC	: Federal Communication Commission
FDM	: Frequency Division Multiplexing
FEC	: Forward Error Correction
FDMA	: Frequency Division Multiple Access
FFBPNN	: Feed Forward Back-propagation Neural Network
FFT	: Fast Fourier Transform
FM	: Frequency Modulation
FPGA	: Field Programmable Gate Array
FSK	: Frequency Shift Keying
GA	: Genetic Algorithms
Gbps	: Giga bit per second
GSM	: Global System for Mobile Communications
GSPS	: Giga sample per second
HARQ	: Hybrid Automatic Repeat Request
HSDPA	: High Speed Downlink Packet Access
Hz	: Hertz
IC	: Integrated Circuit
ICI	: Inter Carrier Interference

IFFT : Inverse Fast Fourier Transform
IR-UWB : Impulse Radio Ultra Wide Band
ISI : Inter Symbol Interference
ITU : International Telecommunication Union
ksps : Kilo sample per second
LNA : Low Noise Amplifier
LPEC : Linear Pattern Envelope Construction
LSE : Link Spectral Efficiency
Msps : Mega sample per second
MBPF : Multi-band Band Pass Filter
MF : Matched Filter
MLCR : Machine Learning Cognitive Radio
MLP : Multi-Layer Perceptron
MSE : Mean Squared Error
MSK : Minimum Shift Keying
MTDE : Multipath Time Delay Estimation
NBI : Near Band Immunity
NTIA : National Telecommunication and Information Administration
OCON : One Class One Network
OFDM : Orthogonal Frequency Division Multiplexing
OFDMA : Orthogonal Frequency Division Multiple Access
OOK : On-Off Keying
OSI : Open System Interconnection
P : Phase
PBC : Pattern Based Communication
PBCS : Pattern Based Communication System
PBMA : Pattern Based Multiple Access
PLL : Phase-Locked Loop
PLLNN : Phase-Locked Loop Neural Network
PM : Phase Modulation
PPM : Pulse Position Modulation
PSK : Phase Shift Keying
PSM : Pulse Shape Modulation
QAM : Quadrature Amplitude Modulation
QoS : Quality of Service
QPSK : Quadrature Phase Shift Keying
RBF : Radial Basis Function
RBNN : Radial Based Neural Network
SCD : Spectral Correlation Density
SCF : Spectral Coherence Function
SDMA : Space Division Multiple Access
SDR : Software Defined Radio
SM : Space Modulation
SNR : Signal to Noise Ratio
SPEC : Sinusoidal Pattern Envelope Construction
SSM : Single Sideband Modulation
SVM : Support Vector Machine
TCM : Trellis Modulation
TDMA : Time Division Multiple Access
ULF : Ultra Low Frequency

V	: Volt
VQ	: Vector Quantization
WAP	: Wireless Application Protocol
WBNN	: Wavelet Based Neural Network
WM	: Wavelet Modulation
WWRF	: Wireless World Research Forum

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LIST OF SYMBOLS

$\vec{a}$	: Pulse shape auto-correlation matrix
$\boldsymbol{A_i}$	: Amplitude (V)
$\vec{b}$	: Transmitted Signal
$\boldsymbol{B}$	: Spectral bandwidth (Hz)
BER	: Bit Error Rate
$\boldsymbol{C}$	: Channel capacity (bps)
Dh	: Depth of SPEC space
$\boldsymbol{d_i}$	: Actual response
$\boldsymbol{e}$	: Absolute error
$\boldsymbol{E_b/N_0}$	: Energy per bit to noise power spectral density ratio (dB)
$\boldsymbol{f_c}$	: Carrier Frequency (Hz)
$\boldsymbol{f_i}$	: Frequency (Hz)
$\boldsymbol{f_c}$	: Central frequency (Hz)
$\boldsymbol{G_i}$	: Glossary matrix
$\boldsymbol{[G_x]}$	: E_b/N_0 value of active glossary
$\boldsymbol{[G_y]}$	: *BER* value of active glossary
$\boldsymbol{H}$	: Number of neuron at hidden layer
H	: Hermetian operator
$\vec{h}$	: Channel coefficients
$\vec{h}_{LS}$	: Channel coefficients for least squared approach
$\vec{h}_{MAP}$	: Channel coefficients for maximum a posteriori
$\vec{h}_{ML}$	: Channel coefficients for maximum likelihood
$\vec{h}_{MMSE}$	: Channel coefficients for minimum mean squared error
$\boldsymbol{I}$	: Number of neuron at input layer
$\boldsymbol{k}$	: Index of channel coefficient
$\boldsymbol{l}$	: Index of tap
LSE	: Link spectral efficiency (bps/Hz)
$\boldsymbol{N}$	: Noise power (W)

$\boldsymbol{Ns}$	: The utilized number of symbols
$\boldsymbol{q}$	: Number of neuron at output layer
$\vec{r}$	: Received signal
$\vec{R}_h$	: Channel coefficient covariance matrix
$\vec{R}_Z$	: Noise covariance matrix
$\boldsymbol{S}$	: Signal Power (W)
$\boldsymbol{S_{i,j}}$	: Signal pattern (j^{th} pattern in i^{th} glossary)
SNR	: Signal to noise ratio (dB)
$\boldsymbol{SP}$	: Number of sub-pattern
$[\boldsymbol{t}]$	: Active glossary matrix
$\vec{z}$	: Noise model
$\boldsymbol{w_{i,j}}$	: Weight
$\boldsymbol{y_i}$	: Calculated response
$\boldsymbol{\alpha_i}$	: Momentum coefficient
$\boldsymbol{\delta_{i,j}}$	: Synaptic weights
$\boldsymbol{\Delta f}$	: Frequency interval (Hz)
$\boldsymbol{\varepsilon}$	: Mean squared error
λ	: Constant for sigmoid function
$\boldsymbol{\mu}$	: Mean
$\boldsymbol{\eta_i}$	: Learning rate
$\boldsymbol{\Phi_i}$	: Phase (°)
$\boldsymbol{\rho}$	: Number of possible routes
$\boldsymbol{\tau}$	: Number of total pattern
ς	: Channel capacity (bits/unit)
$\boldsymbol{\sigma}$	: Standard deviation
$\boldsymbol{\upsilon}$	: Frequency (unit)
$\boldsymbol{\varphi}$	: Radial function
$\boldsymbol{\psi}$	: Activation function

PATTERN BASED COMMUNICATION: AN INTEGRATED MODULATION AND CODING TECHNIQUE DESIGNED FOR ADAPTIVE COMMUNICATION IN ADDITIVE WHITE GAUSSIAN NOISE CHANNEL

SUMMARY

Inefficient utilization of the bandwidth under the spectrum limitations and reduced performance characteristics of the communication lines due to low Signal to Noise Ratio (SNR) are two prominent restrictions that are encountered in data transmission. In this thesis, a new encoding and adaptive modulation technique, Pattern Based Communication (PBC), is proposed for improving the communication performance under high noise suppression. A system, which uses PBC, is defined as Pattern Based Communication System (PBCS) conceptually. PBCS is an adaptive and perceptual communication system proper to use in Cognitive Radio (CR), which is the sustainable solution to above restrictions. The key point of this system is to construct the optimal communication signals, which consist of patterned data that can be recovered by the cognitive receiver, for a successful data transmission even in low SNR. Lowly correlated patterns are chosen in order to form a communication signal (waveform) and called as a glossary, for which Artificial Neural Network (ANN) in the receiver is trained offline. The distorted signals in the communication medium are then recovered by the pre-trained ANN. Additive White Gaussian Noise (AWGN) channel is used to perform the simulation tests in this study. Some pre-tests are also performed for the other channel models (i.e. fading channel model etc.) to define the future works. The result is a band efficient signal transmission capability at zero Bit Error Rate (BER) approximating the Shannon boundary even in low SNR values. Finally, this study uses adaptive modulation technique PBC to satisfy the user expectation for different communication channel parameters such as data rate and SNR value.

This defined system is an alternative candidate for implementing CR design. Moreover, this new communication structure is applicable to various military purposes in terms of application area. Thanks to the robustness to high levels of noise, it can be used to transmit the signal correctly even when altered by the noise created by the jammer. Furthermore, since the glossaries are formed by the user, an external party can not decode the pattern signal without the knowledge of the glossary. This fact allows the system to be used in military encryption applications. High link spectral efficiency capability allows an increased data rate in low frequency ranges, which improves data transfer performance of the submarine communication systems.

The thesis is organized as follows: First, the aim of the study, the general overview of the literature and the target application fields of the proposed system will be

introduced. In section II, the objective of the proposed system, design restrictions for different communication channel models and the innovative point of PBCS will be given in detail. The transmitter and the receiver structures of PBCS will be explained in section III. Next section, the simulation results will be presented, and finally, the conclusion of the report will be drawn.

ÖRÜNTÜ TABANLI İLETİŞİM: TOPLANIR BEYAZ GAUSS GÜRÜLTÜ KANALINDA UYARLANABİLİR HABERLEŞME İÇİN TASARLANMIŞ TÜMLEŞİK KİPLEME VE KODLAMA TEKNİĞİ

ÖZET

İletişim kanallarındaki İşaret/Gürültü Oranı'nın (SNR) etkisi ve bant genişliği kullanımındaki sınır, bilginin iletişim kanalındaki iletim verimliliğini düşürür. Uyarlanabilir kipleme yöntemlerini kullanan algısal ve dinamik yapıya sahip Bilişsel Radyo (BR), kullanılan iletişim kanalındaki veriyi kanal verimliliğini arttıracak şekilde farklı modülasyon yöntemleri arasında geçiş yaparak kiplediğinden belirtilen sorun için çözüm olarak literatürde yerini almıştır. Bu çalışmada yeni bir modülasyon ve kodlama yöntemi (Örüntü Tabanlı İletişim - ÖTİ) tasarlanmış, bu yöntemi kullanan ve kavramsal olarak tanımlanan sisteme de Örüntü Tabanlı İletişim Sistemi (ÖTİS) adı verilmiştir. ÖTİS, iletişim işaretleri için en iyi haberleşme işaret formunu oluşturan ve yapay zekâya dayalı alıcı ile kip çözülmesini gerçekleştiren bir yöntem kullanır. Bu çalışmada belirtilen yaklaşım kullanılarak Toplanır Beyaz Gauss Gürültü kanalında düşük SNR değerlerinde başarıya ulaşılabileceği gösterilmektedir. ÖTİ ile iletişim sinyalleri tanımlandıkları uzayda mümkün olduğunca birbirlerine dik ve ilintisi düşük seçilir. Seçilen bu örüntüler, sözlük adı verilen kümelerde toplanır. Üretilen her sözlük örüntü tanıma katmanı Yapay Sinir Ağı'na (YSA) öğretilir. YSA, çeşitli çevre şartları nedeniyle bozunuma uğramış başlangıç işaret örüntüsünü tanıyarak, bilginin tekrar oluşturulmasını sağlar. Bu çalışmada Toplanır Beyaz Gauss Gürültü kanalında benzetim sonuçları temel alınmış, diğer kanal şartlarında bazı ön testler yapılarak gelecek çalışmalar için altyapı oluşturulmuştur. Sonuç olarak bu çalışma ile belirtilen kanal şartlarında herhangi bir ek modülasyon yöntemine gerek kalmaksızın düşük SNR değerlerinde ve Shannon limiti çevresinde sıfır bit hata oranı ile bilginin transferi sağlandığı gösterilmiştir. Böylece, bu tezde uyarlanabilir kipleme yöntemi ÖTİ'yi kullanarak kullanıcı isterlerine bağlı farklı iletişim veri hızı ve gürültü oranlarında çalışabilen, BR tasarımında kullanılmaya aday ÖTİS tanımlanmıştır.

Veri iletimindeki başarım iki temel parametreye bağlıdır. Bu parametreler, veri iletimi sırasında kullanılan bant genişliği ve kullanılan kanalın veri iletimi sırasındaki işaret gürültü oranıdır. Shannon Limit Teoremi (Shannon, 1948), kanal veri iletim kapasitesinin, C (saniyedeki bit oranı - bps), kanalın İşaret Gürültü Oranı (SNR) ve kullanılan spektral bant genişliğine, B (Hz), direk bağlı olduğunu gösterir. Dolayısıyla bu iki parametre arasındaki bağıntının eniyilenmesi kanal kapasitesinin en üst düzeye çıkarılmasını sağlamaktadır. Başka bir deyişle, birim frekans başına düşen veri iletim hızı (C/B), Bağ Spektral Verimliliğini (LSE) ifade eder. LSE ile SNR değerleri arasındaki bağıntı da Shannon Limiti ile tanımlanır.

Geleneksel yöntemler ile yapılan modülasyon işlemlerinde belli bir bant genişliği kullanıcıya tahsis edilir. Kanal talebinin tektürsel olmaması durumunda veri iletim başarımının verimliliği düşer.

Mevcut modülasyon teknikleri, birim frekans başına düşen veri iletim kapasitesini (LSE) belli bir kanal SNR seviyesinde bit hata oranını (BER) sıfır yapacak şekilde eniyilemektedir. Ancak gelinen noktada, geleneksel modülasyon yöntemleri halen Shannon Limit'inden bir miktar uzaktadır. Bu nedenle, yüksek gürültü baskısı altında (düşük SNR değerlerinde) LSE değerini en yükseğe çekecek şekilde sıfır BER ile veri iletimi ihtiyaç duyulan çalışma alanları arasında yer almaktadır. Kullanılan bant genişliğini eniyileyerek, tektürsel olmayan taleplerde de en yüksek verimliliği elde etmek amacıyla algısal ve uyarlanabilir yapıya sahip Bilişsel Radyo (BR) (Mitola, 1999) kavramı ilk olarak Mitola tarafından ortaya atılmıştır. Bu kavrama göre kullanılan spektral bant ölçülür, kalan boşluklar da farklı modülasyon teknikleri arasında anahtarlama yapılarak en üst seviyeye getirilir. Böylece mevcut iletişim kanalından en yüksek verim elde edilmeye çalışılır. Örüntü Tabanlı İletişim Sistem'i de, "Örüntü Tabanlı İletişim" (ÖTİ) adı verilen, haberleşme sinyalinin üç temel bileşeninin ötelenmesiyle verinin kiplenmesini sağlayan bir teknik kullanarak, BR kavramında olduğu gibi algısal ve uyarlanabilir bir kavramsal yapı önerir. Ancak, çalışmanın amacı uyarlanabilir kipleme kapsamında önerilen ÖTİ yönteminin başarımını ortaya koymaktır.

İletişim işaretleri, ÖTİS'in verici tarafında sinüs tabanlı örüntü oluşturma yöntemi ile inşa edilir. Üretilen örüntülerden, Yapay Sinir Ağ'ının (YSA) tanıma kapasitesini arttırmak amacıyla, tanımlandıkları uzayda birbirlerine en düşük ilintiye sahip örüntüler arasından seçilerek kümelenir. Seçilen örüntülerin kümesine sözlük adı verilir. Her sözlük kullandığı spektral bant genişliği (B), veri iletme kapasitesi (C) ve gürültü seviyesine (SNR) dayanıklılık değerlerine sahiptir. Böylece kullanıcı istemiş olduğu bağ spektral verimlilik (LSE) değerini sisteme tanıtır ve sistem iletişim kanalındaki gürültü seviyesine göre istenen değere en uygun sözlüğü sözlük uzayından çeker ve kullanır. Kullanılan sözlük verinin ek bir modülasyon tekniğine gerek kalmadan kiplenmesini sağlar.

Her iletişim sisteminde olduğu gibi alıcı tarafta karşılaşılan başlıca sorun, iletişim ortamında yansıma ve gürültü etkisiyle oluşan bozunumlar sonucu işaretin eşzamanlama bloğuna duyduğu gereksinimidir. Bu amaçla literatürde sıkça rastlanan eşleme örüntüsü çözüm olabileceği gibi, işaretin başlangıç ve sonunu tanımlayabilecek önceden eğitilmiş ek YSA bloğu da bu probleme çözüm oluşturabilir. Ayrıca lieratürde yerini almış bir diğer çalışma (Hoppensteadt ve Izhikevich, 2000), hem YSA yapısına sahip hem de evre kenetleme döngüsü özelliklerini kullanan eşzamanlama bloğu da bu probleme çözüm olarak önerilebilir. Alıcı tarafta oluşabilecek bir diğer önemli sorun ise, yansımalardan kaynaklanan çoklu örüntü ortaya çıkmasıdır. Bu sorunda literatürde yer alan ve kendini kanıtlamış çalışmalar ile çözüme ulaştırılabilir. Hatta, YSA kullanımı da bu problemin çözümü olabilmektedir (He vd., 1991). Alıcı tarafta, eşzamanlama ve yansımalardan kaynaklanan bozunum belirtilen yöntemler ile giderildikten sonra, veri YSA'ya beslenir. İlgili örüntü bu blok tarafından tanınarak karşılığı olan ikili bit sırası çıktısı elde edilir. Bu çıktı bit sıralayıcı tarafından bitiştirilerek alıcı tarafta başlangıçtaki veriyi oluşturur. Önerilen kipleme yöntemi, bu çalışma kapsamında benzetim ortamında ve toplanır beyaz Gauss gürültü kanal şartlarında değerlendirilmiş, diğer kanal şartları (çok-yolluluk ve Doppler) için literatürde yerini almış çalışmalar ile çözüm önerileri sunulmuştur.

Önerilen sistemin tüm özellikleri bir arada değerlendirildiğinde, gerek özgün bir yöntem ile sözlük inşa edip başlangıçtaki veriyi kiplemesi nedeniyle oluşan şifreleme kabiliyeti, gerekse yüksek gürültü baskısı altında gürbüz davranış sergilemesi nedeniyle askeri amaçlı uygulamalarda kullanılabilir bir yapıya sahiptir. Bir diğer özellik olarak, yapılan benzetim çalışmalarından da görüleceği gibi bağ spektral verimliliğin yüksek olması sebebiyle, ÖTİ, denizaltı uygulamalarında kullanıma uygun bir teknik olarak önerilebilir.

Tezin birinci bölümünde, yapılan çalışmanın amacı, iletişim sistemleri konusunda literatürde yerini almış modülasyon teknikleri ve önerilen sistemin uygulama alanlarına yer verilmiştir. İkinci bölümde, önerilen modelin detayları, farklı kanal şartlarında karşılaşılabilecek tasarım kısıtları ve bilime getirdiği yenilikten bahsedilmiştir. Bir sonraki bölümde, ÖTİS için tanımlanan Alıcı/Verici yapısı detaylı olarak anlatılmıştır. Bölüm 4'te, benzetim çalışmaları ve yarı donanımsal test ortamıyla elde edilen sonuçlar kullanılarak önerilen sistemin başarımı ve literatürde kullanılan benzer yöntemler ile karşılaştırması yapılmıştır. Ayrıca gelecek çalışmalara taban oluşturabilecek bazı benzetim sonuçlarına da çalışmanın sonuç bölümünde yer verilmiştir. Son bölümde ise, yapılan çalışma genel hatlarıyla değerlendirilmiş ve sonuç yorumlarına yer verilmiştir.

1. INTRODUCTION

There are two crucial aspects those define fundamental restrictions in data transmission channels. One of them is the spectral bandwidth limitation and the other is robustness to noise at error free data transmission speeds. In this thesis, conceptually defined Pattern Based Communication System (PBCS) targets to handle these restrictions.

Each communication signal waveform points to a pattern and this thesis proposes a new waveform design technique. Pattern Based Communication (PBC) refers to the private name of this new modulation and coding approach. This thesis aims to design an adaptive communication signal waveform as flexible as possible and to utilize the features of Artificial Neural Network (ANN) in a physical layer to recognize the constructed waveforms (patterns). Since the boundaries of this work is wide, this study only focuses on a design of flexible waveform construction technique and the performance analysis of the system, which is in Additive White Gaussian Noise (AWGN) communication channel. The other design considerations are defined as a future work with the solution advices. Therefore, these solution advices are located out of the scope for this thesis.

PBCS is a conceptual system that involves a new modulation and encoding technique PBC. It adaptively uses collection of lowly correlated signal pattern sets called glossaries. Each glossary in the glossary space is indexed with using two parameters:

1) The transmission capability under the specified noise suppression in terms of Signal to Noise Ratio (SNR) and

2) Link Spectral Efficiency (LSE) (bps/Hz), which consists of the utilized spectral bandwidth (Hz) by the test signal pattern and its data bit rate (bps).

First proposed by the Shannon Limit Theorem (Shannon, 1948), the channel capacity is directly proportional to the spectral bandwidth and SNR. In order to maximize the channel capacity both parameters should be optimized together.

$$C = B \times \log_2\left(1 + \frac{S}{N}\right) \quad \textbf{(1.1)}$$

where S is a power of signal, N is a power of noise, B is a spectral bandwidth and C is a channel capacity to transmit the data.

In PBCS, the transmitter uses the free spectral bandwidth and SNR capacity of the communication channel, dividing the data bit rate to the utilized spectral bandwidth value to come up with the LSE. This derived value and the measured free SNR capacity correspond to a specific glossary in the glossary space. In the receiver, the disarrangements caused by possible delay and reflections in the communication medium are corrected via a synchronization process that puts the patterns in correct order. Then these patterns are fed into the recognition layer, namely ANN. Having been trained for all possible sets of glossaries, the ANN converts the incoming noise added signal pattern into the associated binary code. The information is reconstructed by concatenating these binary codes.

The certain standards define the allowed spectral bandwidth for the communication channel between the transmitter and receiver. The user utilizes this bandwidth to transmit the information from transmitter to the receiver. The optimization provides the user with the way to maximize the spectral efficiency. Historically, the first aim was to use the communication channel to transmit the information from one location to the other. But, in a time, the optimization for the channel capacity takes the important role in this research area. First, an adaptive structure proposed for modulating the information to handle this problem and the most popular one is called as Quadrature Amplitude Modulation (QAM). After that, this technique enhanced by the new concept Software Defined Radio (SDR) (Mitola, 1995) and Cognitive Radio (CR) (Mitola, 1999) sequentially. The latest and advanced technique CR optimizes bandwidth efficiency by sensing the free spectral bandwidth sections and fills these sections by switching among modulation techniques. To perform this approach, it has some additional features compared to adaptive modulation scheme such as *awareness*, *knowledge*, *decision* and *adaptation*. The most important difference than the conventional techniques of CR is the awareness capability.

The existing modulation techniques deal with the capacity problem in terms of the speed of data transmission. They manage to work under specified SNR value, still they are far from the Shannon Limit. Therefore, the communication systems are in need of a new spectral bandwidth efficient modulation technique, which allows data transfer with approximately zero Bit Error Rate (BER) even under higher noise influence.

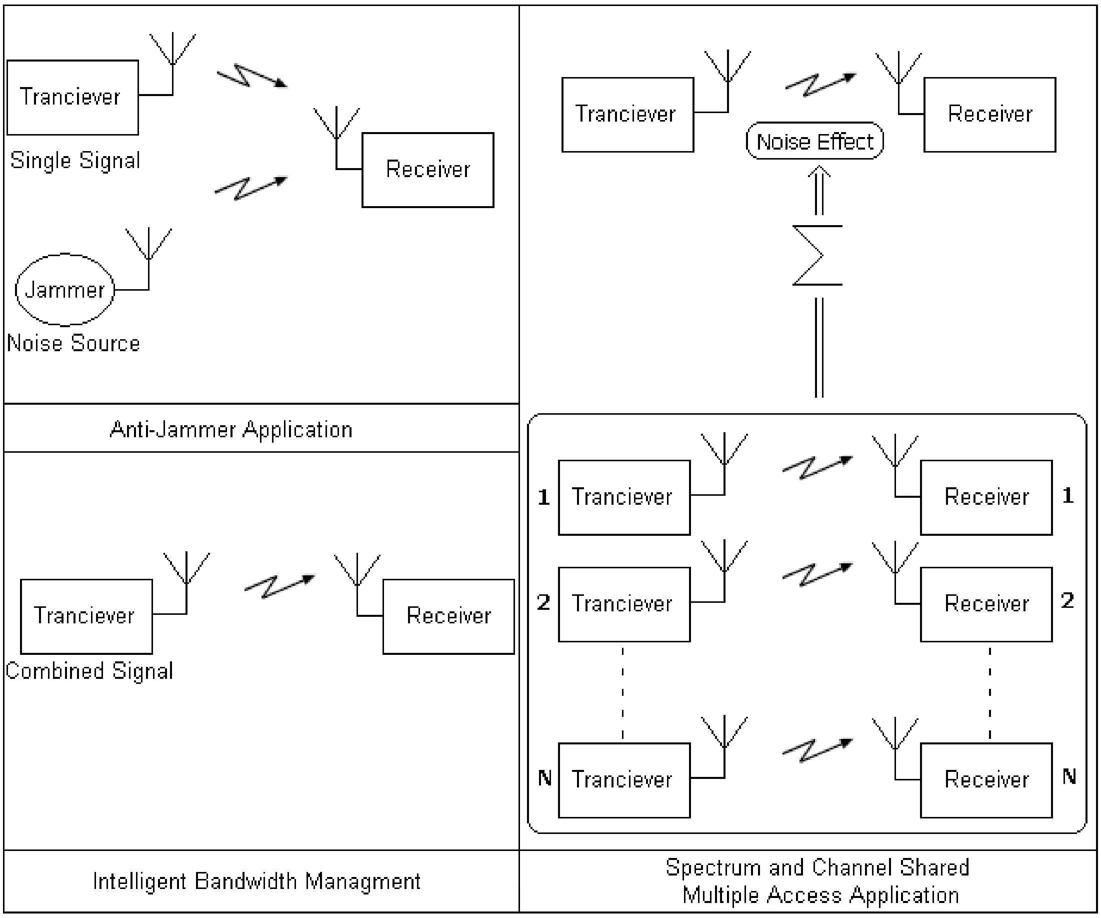

Figure 1.1 : Some application areas of the proposed system.

PBCS uses a new modulation and encoding technique, namely PBC, with adaptive modulation approach. It constructs its own and unique waveforms. Due to this fact of it is applicable to various military purposes. For instance, since the user forms the glossaries, an external party cannot decode the pattern signal without the knowledge of the glossary for peer-to-peer secured communication. Moreover, PBCS can work under high noise suppression. It can be used for transmitting the signal correctly even when altered by the noise created with the jammer. These two facts allow the system to be used in military encryption and intelligent bandwidth management applications. High link spectral efficiency capability allows an increased data rate in low frequency ranges, which improves data transfer performance of the submarine communication systems. The proposed system can separate the information according to the sources, thereof it is proper to use in the multiple accessing applications. Figure 1.1 shows all proposed application areas.

The rest of this section is organized as: Conventional modulation techniques (adaptive and non-adaptive) in the literature are covered in the following subsection. The multiple accessing techniques are introduced. Then the fundamental concepts of cognitive radio are explained such as adaptive modulation, cross-layer architecture and software defined radio. Finally, the concepts of cognitive radio and cognitive network are given to locate PBCS in wireless communication world properly. As a result, adaptive modulation and coding concept is directly related to the proposed modulation technique PBC and the conceptual proposed system PBCS is defined as a candidate to construct cognitive radio in this thesis. These relations are described in this thesis.

1.1 Modulation Techniques

Bandwidth demand for digital communication rapidly rises due to evolution of new generation computational devices and dramatically increasing number of users. This increase in demand directs us to use the frequency spectrum as the physical resource. The efficiency of bandwidth utilization takes an important role in spectrum management and SNR has an important role for bandwidth limitation. Bruce Fette (2006) says that “the frequency spectrum is the lifeblood of communication system, it should be managed well to obtain better performance on transmission line”. Shannon (1948) first formulated the theoretical limit of the data bandwidth with respect to frequency bandwidth. After this milestone, thousands of studies have performed until today, in order to increase the bandwidth efficiency.

Many modulations and coding methods have taken place in the literature. The initial techniques have based on fundamental analog modulation methods such as Amplitude Modulation (AM), Frequency Modulation (FM) and Phase Modulation (PM). Some methods have derived from them such as Single-Sideband Modulation (SSM), Space Modulation (SM) etc. Subsequently, the communication world has tended to digital modulation. The simplest way of digital modulation is called as On-Off Keying (OOK). Well-known methods in digital modulation literature depend on “shift keying” of the signal features. These benchmark studies shift one of the features of communication signal to present “0” and “1” bits of digital data. Many digital modulation methods are developed with this logic. The initial methods are called Amplitude Shift Keying (ASK), Frequency Shift Keying (FSK) and Phase Shift Keying (PSK). For instance, the frequency of RF signal is changed between two different values in FSK. Figure 1.2 explains this process.

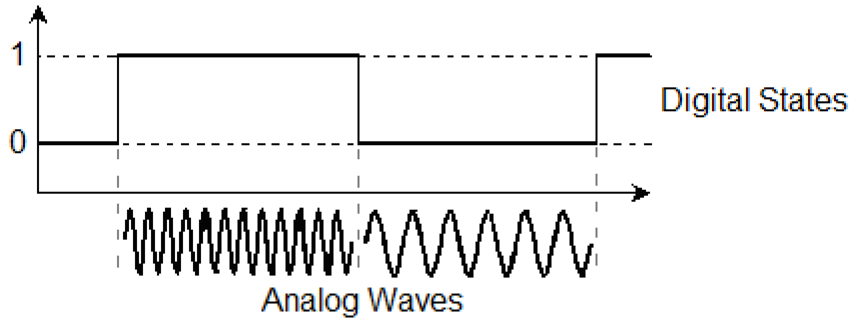

Figure 1.2 : Frequency shift keying (FSK).

Many derivative digital modulation techniques are developed such as Quadrature Amplitude Modulation (QAM - a combination of PSK and ASK), Continuous Phase Modulation (CPM) (Minimum Shift Keying(MSK), Gaussian MSK and Continuous Phase Frequency Shift Keying(CPFSK)), Pulse Position Modulation (PPM), Pulse Shape Modulation (PSM), Wavelet Modulation (WM), Trellis Modulation (TCM) and Orthogonal Frequency Division Multiplexing (OFDM). One of the different studies for developing digital modulation technique in literature has used vector modulation techniques (Oetting, 1979). Another method has utilized spatial distribution of the capacity into shorter propagation ranges as in the case of cellular systems. The initially developed modulation techniques were mentioned above. Still some of them are in use communication systems. On the other hand, the increase in the number of users at communication line appears as another issue, therefore multiple-access sharing the channel in time, frequency, code or space (Brand and Aghvami, 2002) is the strategy that allows better use of common spectrum.

1.2 Multiple Accessing Techniques

The transmission of the information with desired data rate is limited by the spectral bandwidth restriction and the increase in the number of users. The assigned frequency band for each user should be utilized by the best way. There are many studies about multiple accessing in the literature for this purpose. The main goal behind these studies is to provide the best channel conditions and to obtain the maximum data rate on the transmission line for each user.

The well known multiple accessing techniques can be listed as Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), Code Division Multiple Access (CDMA) and Space Division Multiple Access (SDMA) (Zigangirov, 2004, Steven, 1999, Finkenzeller, 2003). Additionally, many derivative techniques were developed from these benchmark methods. For instance, Weber et al. (2002) first proposed the combination of TDMA and CDMA. This technique is named as Time Division CDMA. Benjamin et al. (2003) proposed the multiple antennas usage for multiple accessing. However, one of the innovative methods in the multiple accessing techniques has taken the most important place in the literature. It is called Orthogonal Frequency Division Multiple Access (OFDMA). The benchmark studies in the multiple accessing applications are given below in detail.

1.2.1 Frequency division multiple access

This technique assigns a unique channel to individual users. The spectral bandwidths of channels are relatively narrow, so this technique works as narrow band communication. If the user is not online, the assigned channel is idle. This situation causes reduction of the performance of transmission line (Zigangirov, 2004). This technique is shown in Figure 1.3.

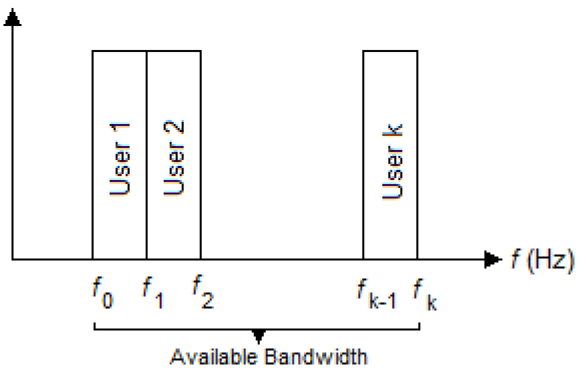

Figure 1.3 : FDMA frequency utilization scheme.

1.2.2 Time division multiple access

This technique divides the transmission time into many time slots. Only one user utilizes each time slot. TDMA shares a single carrier frequency with several users, where each user makes use of non-overlapping time slots. Analogously to FDMA, if a channel is not in use, the corresponding time slots sit idle and cannot be used by any other users. This technique is more complex than FDMA and it needs to use digital modulation (Zigangirov, 2004). Figure 1.4 shows this technique.

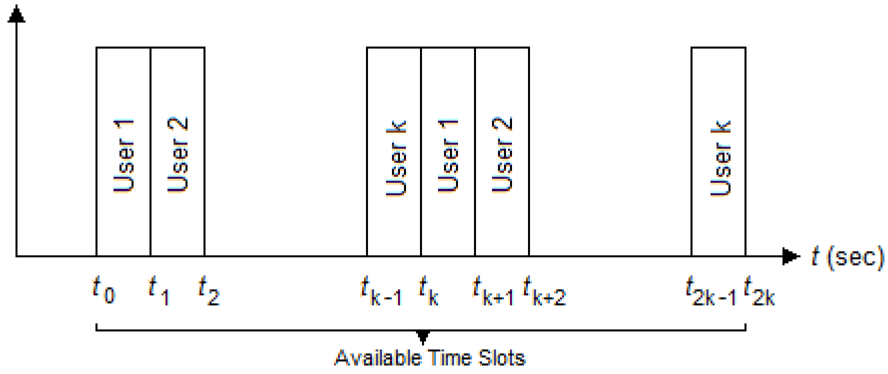

Figure 1.4 : TDMA time utilization scheme.

1.2.3 Code division multiple access

This technique has completely different approach than FDMA and TDMA. CDMA systems do not attempt to allocate disjoint frequency or time resources to each user. Instead of that, the system allocates all resources to all active users simultaneously. There are many versions of this technique. The basic approach is called direct sequence (DS) CDMA. This system has the narrowband message signal. This signal is multiplied by very large-bandwidth signal named the spreading signal. All users in a DS-CDMA system use the same carrier frequency and time slot. Each user has its own spreading signal, which is approximately orthogonal to the spreading signals of all other users. The receiver performs a correlation operation to detect the message addressed to a given user. Signals from other users appear as noise due to de-correlation. The receiver requires the spreading signal used by the transmitter for detecting the message signal. Each user operates independently without knowledge of the other users. Potentially, CDMA systems provide a larger radio channel capacity than FDMA and TDMA systems. The basic idea is shown in Figure 1.5 (Zigangirov, 2004).

The signals are representing by “-1” and “1” at CDMA structure. In below example, two users are located. The signals have the following information: User 1 (1,-1,-1,1) and User 2 (-1,1,-1,1). Figure 1.5.b shows the summation of these signals. Indeed, this combined signal cannot be separated at this format. This system uses spreading signal, which has generally lower period time. Its period is selected around ¼th of original signal's period. Total signal and the spreading signal are multiplied. Figure 1.5.d represents the new signal forms. Figure 1.5.e shows the summation of these signals. The combined signal form is transmitted through communication channel. This signal form can be separated at the receiver side by multiplying with the spreading signals. Figure 1.5.f shows the separated signal forms. It shows the same form as the original signal.

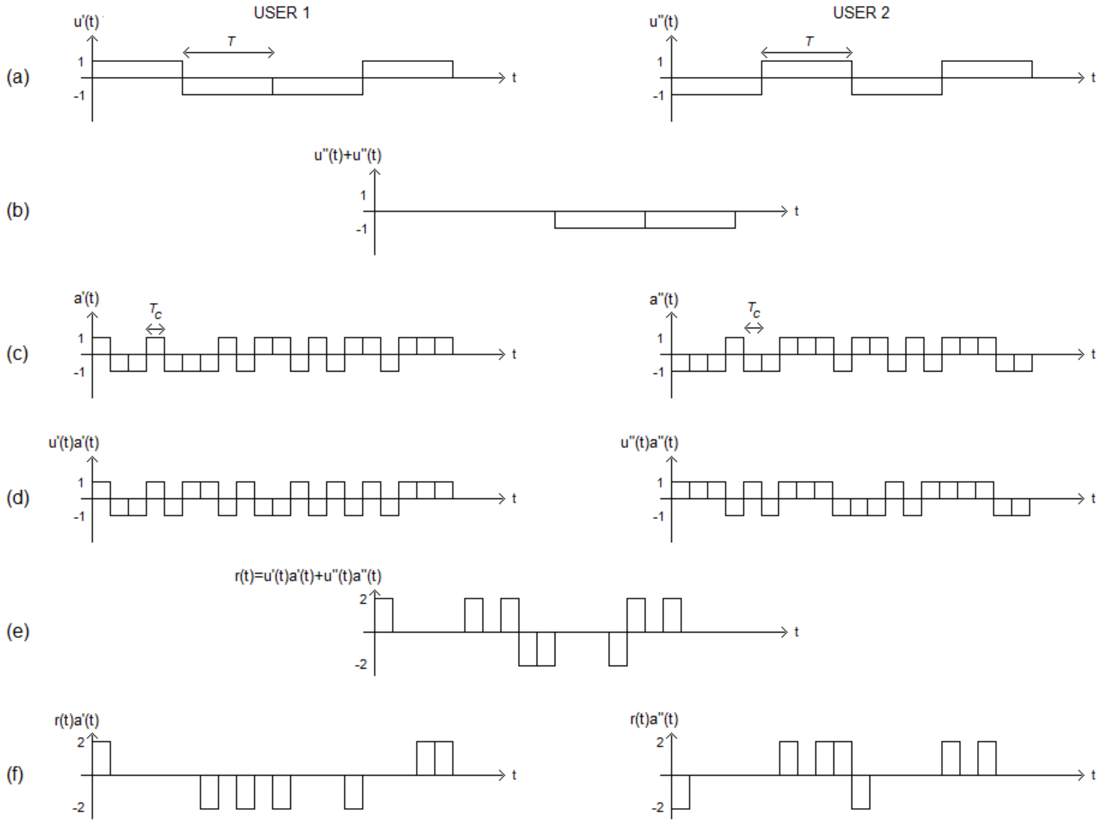

Figure 1.5 : CDMA structure.

1.2.4 Orthogonal frequency division multiple access

One of the popular techniques is OFDMA (Fazel and Kaiser, 2003). This technique descends from FDMA but it has some innovative points. OFDMA is a multi-user version of the popular OFDM digital modulation scheme that will be explained in section 2. Multiple accesses are achieved in OFDMA by assigning subsets of subcarriers to individual users. It allows simultaneous low data rate transmission from several users. The advantages of OFDM are the combating multi-path with less complexity and the achieving higher spectral efficiency.

These methods have strong positions in multiple accessing applications. Many studies in the literature are performed for comparisons them in terms of the channel capacity (Rojas et al. 1998, Bisaglia et al. 2005, Krishnamurthy et al. 2003, Ma et al. 2006) and the spectral efficiencies (Moeneclaey et al. 2001, Cho 2004, Hammuda 1994). Because of these methods are recognized as standards in the communication world, some studies have evaluated the usage of these conventional techniques for each communication generations (Fan, 2006).

The utilization of the conventional modulation and multiple accessing techniques is continuing in communication systems. However, these techniques still depend on the fixed frequency spectrum utilization. Mitola has defined Cognitive Radio concept in the mid of 1990's. The concept depends on adaptive modulation structure with the awareness of radio link spectrum. This idea has opened horizons to the researcher specialists about dynamic frequency spectrum utilization. Therefore, the way to go cognitive radio will be pointed out systematically at the next subsections.

1.3 Adaptive Modulation and Coding

Adaptive Modulation and Coding (AMC) or link adaptation is the method that switches the signal modulation, coding and other protocol parameters according to the real-time conditions on the radio link. Adaptive modulation systems aim to improve the data rate transmission and/or bit error rates by exploiting the channel information that is present at the transmitter.

High Speed Downlink Packet Access (HSDPA) will be taken an example to explain AMC. HSDPA switches two parameters to obtain the optimum performance in the radio link. These parameters are the modulation types (QPSK, 16-QAM) and code rate changed by Forward Error Correction (FEC) method. QPSK is the derivative form of PSK and called Quadrature Phase Shift Keying. It is more robust and can tolerate higher levels of interference than 16-QAM, but has lower transmission bit rate. In opposite, 16-QAM is a derivative form of QAM (Quadrature Amplitude Modulation) and it has twice-higher bit rate but is more prone to errors due to interference and noise than QPSK. Hence, it requires stronger FEC coding to provide more redundant bits and lower information bit rate. The FEC code utilization has a rate of 1/3 in HSDPA, but it can be varied effectively by bit puncuring and Hybrid Automatic Repeat Request (HARQ) with incremental redundancy. HARQ is a combination of FEC coding and error detection using the automatic repeat request error-control method. In proper radio link conditions, more bits can be punctured and the information bit rate is increased. In noisy channel conditions, the system transmits all redundant bits and the information bit rate drops. In very bad link conditions, retransmissions occur due to HARQ, which ensure correct reception of the sent information but further slow down the bit rate. Thus, HSDPA adapts to achieve very high bit rates on clear channels using 16-QAM and close to 1/1 coding rate (14 Mbps). In noisy channel conditions, HSDPA adapts to provide reliable communications using QPSK with 1/3 coding rate but the information bit rate drops to about 2.4 Mbps. This adaptation is performed up to 500 times per second.

On the other hand, PBC constructs different glossaries with its innovative waveform envelope construction algorithms. Each glossary has different features and modulation levels. This means that each glossary is robust against to the different level of interference and varying data rate capability. Control unit manages the active glossary (waveform) in the time slots according to the real-time radio link conditions. The aim of this structure is to provide the optimized radio link conditions as all AMC approaches. Take into account, PBC satisfies the AMC requests with waveform construction algorithms and control unit in its structure. One of the well-known modulation techniques, QAM, is potential competition with PBC modulation approach. Therefore, the relationship between QAM and PBC will be discussed in section 2 in terms of adaptive structure and they are compared in simulation platform. The results are given in Section 4.

1.4 Architectures in Communication Systems

There are two different architectures defined in communication systems. These are traditional layered architecture and cross-layer architecture. The “Signal data” path in Figure 1.6 corresponds to the traditional layered architecture. Any layer in this architecture only communicates with its upper and lower layers. On the other hand, the cross-layer architecture includes “Cognitive Engine” that communicates more than one layer simultaneously. It takes the information from different layers and utilizes it to optimize the system performance by managing these layers for the real-time radio link conditions.

Conceptual system PBCS has a block, which includes “Glossary Space”, “Glossary Selector” and “Spectrum Sense” blocks. This block corresponds to “Control Unit” in Figure 1.6. It is located on the Data Link layer and only communicates with Physical layer block. It uses glossary information and manages the active glossary in time slot to find optimum solution in terms of data rate. PBCS has traditional layered architecture with this structure. It can be extended with some additional features and designed for cross-layer architecture.

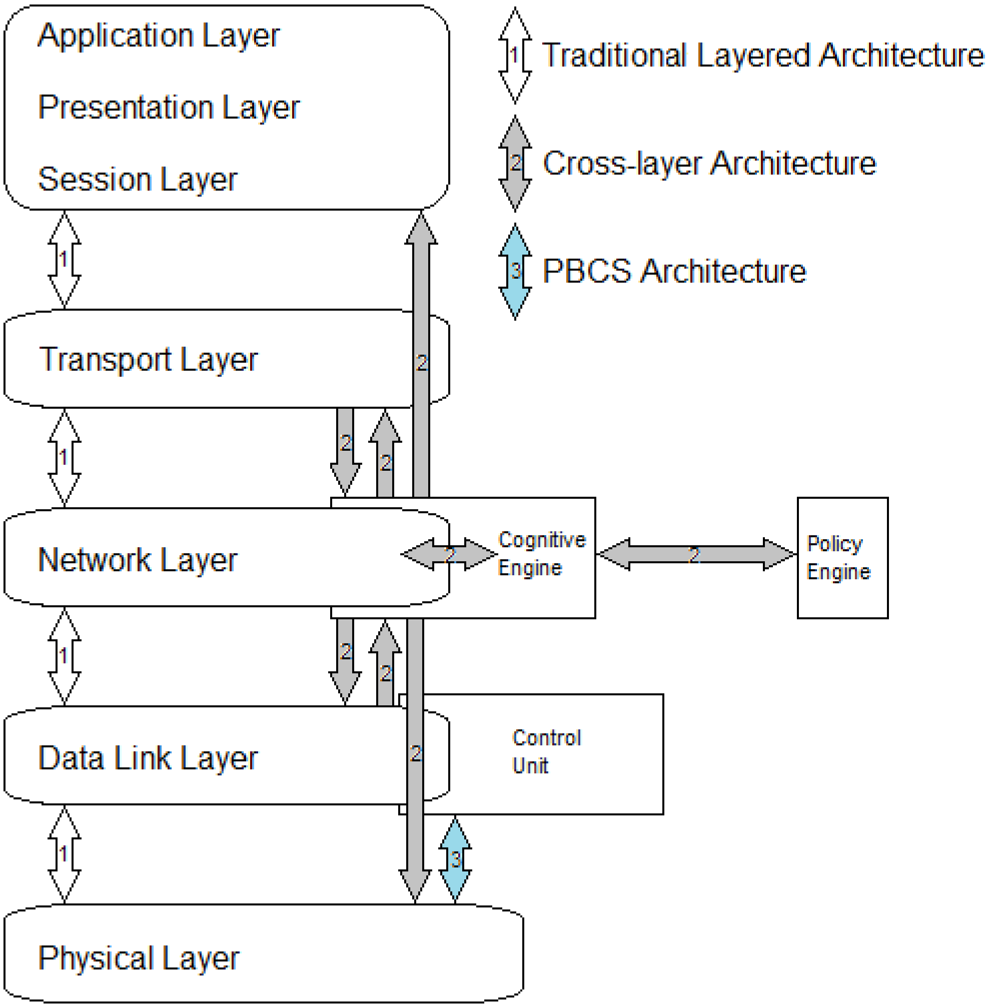

Figure 1.6 : Architectures of communication systems and the position of PBCS.

1.5 Software Defined Radio

A Software Defined Radio (SDR) is a flexible radio communication system. Mitola (1995) first introduced this system. A basic SDR system consists of a personal computer equipped with any proper analog-to-digital converter, and RF front end. The general-purpose processor accomplishes significant amounts of signal processing load for processors. On the other hand, SDR can receive and transmit widely different radio protocols (sometimes referred to as a waveforms) based solely on the software used. In this case, the design needs to use some specific processors designed for SDR. The important point of this structure is that SDR has not any intelligent layer. The external control unit should be installed to control it. SDR was the first hardware platform of adaptive modulation and coding structure. Thereof, software radios have significant utility for the military and cell phone services, both of which must serve a wide variety of changing radio protocols in real time. However, it was not as capable as cognitive radio to utilize the spectral bandwidth dynamically. Since it has not any awareness capability and cognitive engine, it needs to get external support.

1.6 Cognitive Radio

SDR concept has matured with Cognitive Radio (Mitola, 1999) approach. This idea depends on the utilization of the frequency spectrum, dynamically. The dynamic utilization of the spectral bandwidth is performed with the software flexibility. Mitola has described the concept of Cognitive Radio circle.

According to this approach, six states in this system are defined. “Observe” is doing observation and sensing spectral situation on the communication medium. The output of state “Observe” is fed into the “Learn”, “Decide” and “Orient” states. State “Orient” manages the system. It decides the next state with the received information from “Observe” and “Learn”. The alternative next states are “Act”, “Plan” and “Decide”. “Plan” works with “Learn” and provides the information to “Decide”. “Orient”, “Plan” and “Decide” states provide the adaptation and awareness abilities to the system. Moreover, “Learn” state presents the knowledge feature of CR circle. “Orient”, “Learn” and “Decide” give their outputs to the state “Act” and it transmits the information to the communication medium. This chain always realizes in a loop. This structure has four important features such as *awareness*, *knowledge*, *decision* and *adaptation*. However, the awareness gives the cognition capability to the CR. The relation between SDR and CR will be discussed in the further section.

CR is expected by proponents to become the dominant technology in radio communication after the beginning of 2000’s. Different standards are defined CR with small differences. These are listed by the alphabetic order at below.

- *Alliance for Telecommunications Industry Solutions (ATIS)*: A radio that
    - monitors its own performance,

  - monitors the link quality through sounding or polling,
  - varies operating characteristics, such as frequency, power, or data rate,
  - uses closed loop action to optimize performance by automatically selecting.

- *Federal Communications Commission (FCC)*: CR is a radio that can change its transmitter parameters based on interaction with the environment in which it operates.
- *International Telecommunication Union (ITU)*: A radio or system that senses, and is aware of, its operational environment and can dynamically and autonomously adjust its radio operating parameters accordingly.
- *National Telecommunications and Information Administration (NTIA)*: A radio or system that senses its operational electromagnetic environment and can dynamically and autonomously adjust its radio operating parameters to modify system operation, such as maximize throughput, mitigate interference, facilitate interoperability, access secondary markets.
- *Wireless World Research Forum (WWRF)*: Cognitive Radio employs a dynamic time-frequency-power based radio measurement and analysis of the RF environment, to make an optimum choice of carrier frequency and channel bandwidth to guide the transceiver in its end-to-end communication, with quality of service being an important design requirement.

According to these definitions, SDR is a flexible platform for switching between different modulations and coding methods. If the system has some additional capabilities such as having awareness, creating knowledge, making decision and adaptation, this system is called CR.

Many studies about CR appeared in the literature, in a relatively short time. CR techniques opened a new direction of the efficient spectrum management (Jondral, 2005). Cognition in radio communication can be considered as the utilization of the frequency bandwidth in an adaptive manner with respect to channel availabilities. Increasing performance and cost efficiency of the digital computational units allow software defined radios to implement cognitive strategies in communication.

The spectrum management techniques have met with the new concept in the late 1990s. Due to the big step in die producing technology of Integrated Circuits (IC), Digital Signal Processors (DSP) is begun to use nearly all telecommunication systems at these years. This situation enabled the designers to come up with flexible radio design. This technology is called Software Defined Radio. The semiconductor vendors of this technology have made application layer software in the early 2000's. Similarly, in the late 2000's, many radios will have the software functionality for supporting user based radios. This initiate the generation of CR implementation (Arslan, 2007).

The fast development in the hardware stage triggered the implementation of new software based techniques such as SDR and CR on the communication world. Some well-known modulation techniques are used by them to improve the communication performance. Two of them could be directly related to this study. These are OFDM and Pulse Shape Modulation (PSM). The relationship between OFDM, PSM and PBC will be discussed in section 2 in terms of applicability in CR concept.

1.6.1 Artificial intelligence in cognitive radio

Cognitive radio has provided a wide range of technologies for making wireless systems more flexible via transceiver platforms. It enhanced computational intelligence (Arslan, 2007). Some studies towards context aware services are located in the literature. In one branch of study, artificial intelligence concepts are applied to develop perception, planning and machine learning techniques. The other branch is extensible markup language. Mitola's book explains the second branch with all details (Mitola, 2006). The proposed technique in this thesis is close to the first branch of these researches in terms of concept. It uses artificial intelligence and machine learning techniques to construct a communication system under CR approach.

Cognitive radio designer mainly does not propose a new modulation and/or encoding technique. The system uses the conventional methods. The main idea behind cognitive radio concept is that the state of frequency spectrum is identified first, and then the most proper modulation technique is selected for obtaining the maximum transmission performance. The system switches the modulation methods to obtain the best performance in communication line in terms of the channel capacity. This structure based on four important features such as adaptation, knowledge, decision and awareness. CR was defined in one survey (Akyildiz et al., 2006) as "a CR is a radio that can change its transmitter parameters based on interaction with the environment in which it operates".

Another description for cognitive radio in the literature is placed as a paradigm for wireless communication in which either a network or a wireless node changes its transmission or reception parameters to communicate efficiently without interfering with licensed users. This alteration of parameters is based on the active monitoring of several factors in the external and internal radio environment, such as radio frequency spectrum, user behavior and network state.

The main purpose of CR concept is to use the spectral bandwidth in an optimized way. CR uses the most proper modulation technique to obtain the best performance. Artificial Neural Network (ANN) is the well-known classification method in the literature. Due to this fact, the utilization of ANN in CR is a very logical way to construct an optimized communication system. Louis and Sehier (1994) first proposed this combination; they have asserted the concept of neural network utilization in recognition of modulation. In this study, they have compared some classification algorithms for this aim, and showed the best solution. Azzouz et al. (1996) focused on neural network usage to identify analogue/digital modulation types. They have defined a global procedure for recognition of analogue and digital modulation types. Researchers have concluded that 10dB is a lower bound of SNR to determine the modulation type in this study. Kim et al. (2003) investigated the applicability of hierarchical neural network for recognition of modulation type. They have realized this system with DSP from Texas Instruments (TMS320C6701) for the identification of 11 different modulation types. This study emphasizes that the hierarchical neural network classifier has better performance than the conventional classifiers. The other two studies in the literature (Yaqin et al., 2003, Arulampalam et al., 1999), have worked on the optimization of neural network design. They have found out the best performance of the modulation recognition.

1.6.2 Spectrum sensing and artificial intelligence

A prominent subject under the CR concept is spectrum sensing, for which extensive work is done in the literature. This work includes Matched Filtering (MF) (Turin, 1960), Waveform-based sensing, Cyclostationary-based Sensing, Energy Detector-based Sensing and Radio Identification (Arslan, 2007). MF is the well known error recovery and supporting method for the conventional modulation technique. PBC is an alternative approach to MF in AWGN channel in terms of the robustness against to noise. The most significant disadvantage in the practical use of MF is the information need about bandwidth, operating frequency, modulation type and order, pulse shaping, frame format etc. at the receiver side. The proposed system overcomes this disadvantage, since the information is embedded into the recognition process of ANN.

Various methods utilize ANN to perform spectrum sensing. These approaches prove the ANN capability for spectrum sensing and they support the idea behind our approach. Therefore, all of these studies will be evaluated with their innovative points in this section.

Fisher and Westover (2003) have published related study with our work in literature. The proposed structure in this study has some similarities with our proposed method, but the main idea behind our study is completely different. The authors propose the multiple accessing technique with intelligent switch named neuron. They combined signal with special technique. The neuron allows the different information via each channel sequentially with intelligent way as a referee. The neuron in this system works like that CPU in the computer systems and it increases the efficiency of channel utilization.

Palicot and Roland (2003) have successfully managed to determine the bandwidth and its shape by using RBNN (Radial-based Neural Network). This study proved that a self-adaptive terminal using for blind recognition is a realistic approach. In another study, Gandetto et al (2004) have used Feed-Forward Back Propagation Neural Networks (FFBPNN) and Support Vector Machine (SVM) with Radial Bases Function (RBF) to make time-frequency analysis. The authors have shown that the neural classifiers provide the possible solution to identify blind radio configuration with this study.

Fehske et al. (2005) have used Spectral Correlation Density (SCD) and Spectral Coherence Function (SCF) to evaluate the performance of Multi-layer Perceptron (MLP) for spectrum sensing. MLP is used in order to classify the different modulation methods (BPSK, QPSK, FSK, MSK, and AM). The authors have shown that the signal modulation types can be determined reliably with MLP. Another related work with Fehske's study is performed in the same year. Le at al. (2005) have used five different standard deviation of the signal (Amplitude, Envelope, Phase, change in Phase and absolute change in Phase) to reach modulation decisions. The authors selected neural classifier OCON (One Class One Network). The researchers performed the proposed method on different modulation techniques (AM, FM, BPSK, QPSK, 8-QAM and 16-QAM). They have shown that the simple MLP can be used for classifying all existing/overlapping signals. This classification method provided better dynamic spectrum management opportunity in CR systems. The same researchers have expanded this study with using complex quasi-band for signal feature extraction in (Le at al., 2006). They have shown whole design challenges with details and the availability of neural classifier usage for this purpose.

Clancy and Stuntebeck (2007) have performed another study. They concerned with machine learning techniques on CR. Estimating channel statistics is used for the learning engine stage of the proposed system. The objective function is optimized by the different machine learning methods (Neural Networks, Evolutionary Strategies and etc.). The authors have tried to formalize the learning architecture behind CR. They have defined the selection method for the most appropriate machine learning application for dynamic spectrum management.

Tsagkaris et al (2008) have introduced and evaluated the artificial neural networks based learning schemes in cognitive radio. The aim of this study is to predict the data rate capabilities of a specific radio configuration. The expectation of this study is to assist a cognitive radio system. The technique compares entire candidate radio configurations in terms of physical layer. It selects the best alternative to operate.

Baldo et al. (2009) have presented an implementation of a cognitive controller for dynamic channel selection in IEEE 802.11 wireless networks. This controller has a neural network based engine to learn and predict the performance achievement on different channels. Performance evaluation carried out on a real wireless network deployment. It demonstrated that the proposed cognitive controller could effectively learn how the network performance is affected by the changes in the environment. It also performed dynamic channel selection thereby providing significant throughput enhancements when compared to conventional channel assignment strategies. Due to the results in this work, the simulation tests are performed for IEEE 802.11 wireless network features in the last part of thesis.

Chen et al. (2010) have proposed a new concept Machine Learning Cognitive Radio (MLCR). This system extracts the signal features from its waveforms received through various radios. The user of this system assigns the states (signal, action) to each waveform. The MLCR uses ANN to learn and classify the signal states depends on their features. The signal and action pairs are stored in the knowledge base. These pairs can be retrieved by MLCR automatically. The result proves ANN recognition capability through communication signals.

1.7 Cognitive Network

Cognitive Network (CN) is a different approach of data network that helps to meet different technologies from several research areas such as machine learning, knowledge representation, computer network, network management. The aim of CN is to solve some problems defined in the current networks.

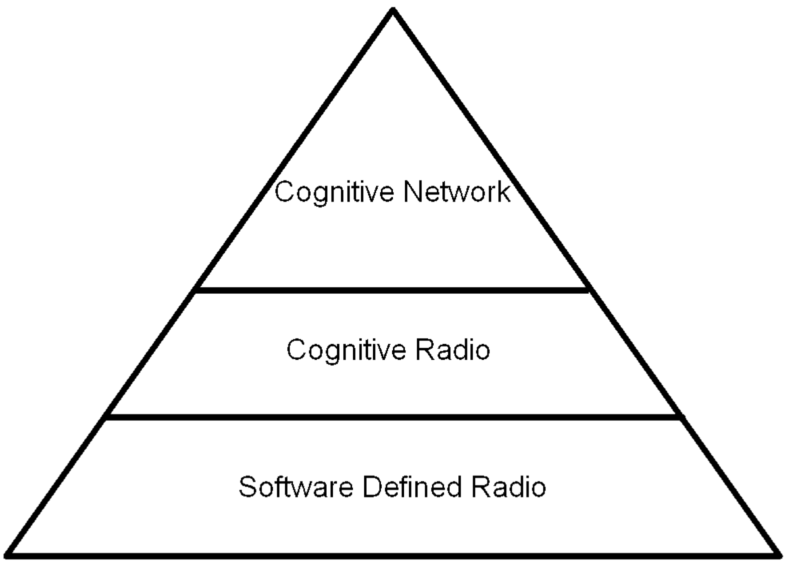

Figure 1.7 : The relationship between CN, CR and SDR.

The relation between Cognitive Network, Cognitive Radio and Software Defined Radio is given in Figure 1.7. SDR is the fundemental form of high efficient and flexible radio system. SDR system or device implements operating functions but could not control it. This means that SDR is not an intelligent system but the base of the flexible radio communication structure. CR requires a flexible radio device hence

SDR is an ideal base platform for it. CR is highly intelligent radio systems with multiple functionalities and efficient resource allocation. It has some additional features such as having awarness, creating knowledge, making decision and adaptation. Therefore, CR can be located to the upper layer of SDR. The upper layer of CR is called CN that covers all the layers of the Open System Interconnection (OSI) model.

Thomson et al. (2005) define Cognitive Network as “A network with a cognitive process that can perceive current network conditions, plan, decide, act on those conditions, learn from the consequences of its actions, all while following end-to-end goals”. This cognition loop senses the environment, plans actions according to input from sensors and network policies. It decides the scenario, which satisfy the best end-to-end purpose with using a reasoning engine, and finally acts on the chosen scenario. The system learns from the past (situations, plans, decisions, actions) and uses this knowledge to improve the decisions in the future. It means that CN has an artificial intelligence feature in its structure.

The definition of Cognitive Network in (Thomson et al., 2005) only describes the cognitive loop and adds end-to-end aims that would distinguish it from CR. It does not mention the knowledge of the network. This definition seems to be incomplete since it lacks knowledge which is an important component of a cognitive system as discussed in the other related studies such as Clark et al. (2003), Mahmoud (2007), Mitola (2000) and Balamuralidhar and Prasad (2008).

Fortuna and Mohorcic (2009) filled this gap with the new definition. According to this approach “Cognitive Network is seen as a communication network augmented by a knowledge plane that can span vertically over layers (making use of cross-layer design) and/or horizontally across technologies (learning techniques, decision making techniques, etc.) and nodes (covering a heterogeneous environment)”.

Take into account, SDR is the flexible radio system for the base of adaptable wireless communication system. CR is the adaptable and intelligent radio system based on SDR. Finally, CN is the term of communication network that has adaptable and intelligent capabilities. It basis on CR and solves the problems at the traditional communication networks. CN covers all layers in OSI model and optimizes the overall performance of the communication system.

On the other hand, PBC is an alternative modulation and coding technique by using Amplitude Frequency Phase Shift Keying (AFPSK). This approach will be given in section 3. AFPSK provides PBCS with a flexible and adaptive capability in terms of information modulation and coding. Thus, it is a kind of AMC method. In addition to this, its structure is proper to extend for cross-layer architecture. The control unit of PBCS locates in data link layer and communicates with the external glossary space. Control unit of PBCS manages the system optimization. Thereof, PBC is a kind of AMC and the conceptual system, namely PBCS, is a candidate for locating at CR in Figure 1.7. However, PBCS is not a cognitive network with this structure. When the multiple accessing capabilities will be defined and implemented, it can be called cognitive network. The extension of control unit to cognitive engine is defined as a future work in this study and the aim of thesis is determined that the performance of new AMC technique in AWGN channel.

Inspired by the recognition capability of the human, PBCS follows perceptual selectivity of a human being. When the human recognizes the sounds, the brain keeps perceiving sounds in the native language while shutting down the others. The trained brain can focus on the native language of the human being among all other languages spoken at the same time. In PBCS, the communication system selectively recognizes and recovers the communication signal into known patterns. Just like the human brain, it ignores the unknown patterns. This mainly depends on neural cognition of the biological systems (Haykin, 2005). These systems use glossaries formulated as probabilistic continuation rules (Andras, 2005). The idea behind this study is teaching the appropriate communication patterns to the cognitive receiver. It can directly recover and recognize without any additional demodulation, decoding and error recovery operations. This feature allows even separation of multiple signal patterns in a common frequency channel.

Adaptive modulation concept and communication signal pattern recognition capability of the neural network systems have combined in our approach (Ustundag, 2010). The prior information in adaptive receivers besides the conventional function based decoding abilities. This proposed technique uses neural network approach for decoding. Due to the neural network capability of separating existing/overlapping signals, PBCS can be used for single user systems. It is called Pattern Based Communication (PBC). Alternatively, it can be adapted to multi-user systems. It is called Pattern Based Multiple Access (PBMA). This work focuses on single user system. PBMA is at the out of scope for this work. It is defined as future works with cognitive network implementation. Next section contents the model of PBCS.

2. PATTERN BASED COMMUNICATION SYSTEM

A major objective of digital wireless communication systems is to provide the maximum error-free data transfer rate under the spectral bandwidth limitations and the different channel noise levels. Pattern Based Communication System (PBCS) targets to optimize the data transfer rate under these restrictions with its own modulation and coding technique called Pattern Based Communication (PBC). First subsection contents the objective of PBCS and mathematical expression of the optimization problem. The most common and proper methods for PBC in adaptive modulation and Cognitive Radio (CR) will be discussed in the second subsection. Although this thesis is related to the Additive White Gaussian Noise (AWGN) channel and baseband applications of the proposed modulation technique, the design considerations for the implementation of PBCS in RF communication are explained in the section 3. The location of PBCS in wireless communication world is given at the end.

2.1 The Objective of PBCS

The aim of PBCS is to provide the maximum data rate under the communication channel restrictions. The overall performance of communication system can be evaluated by four main parameters. These are the spectral bandwidth utilization (B in Hz), the Signal to Noise Ratio (SNR in dB), the channel capacity (C in bps) and the Bit Error Rate (BER). The concept of Link Spectral Efficiency (LSE in bps/Hz) is derived from B and C. It consists of the utilized spectral bandwidth (Hz) by the test signal pattern and its data bit rate (bps). The channel noise level directly affects the transmission capacity of channel. It refers to SNR. All cognitive communication systems target to maximize LSE under the specified noise suppression in terms of SNR at approximately zero BER.

The concept of Energy per bit to Noise Power Spectral Density Ratio (E_b/N_0) contents the relation between SNR and LSE. The following equation shows this function.

$$\frac{E_b}{N_0} = \frac{S/C}{N/B} = \frac{S/N}{C/B} = \frac{10^{SNR/10}}{LSE} \qquad \textbf{(2.1)}$$

where, S is the total signal power and N is the total noise power in the communication channel. The optimization of E_b/N_0 is a direct consequence of the optimization of SNR over LSE with the minimized BER. The constraint of the objective function is average BER. The objective function and the constraints of the single object optimization problem are given in equation 2.2.

$$\begin{array}{ll} \text{Minimize} & f = \sum_{i=1}^{n} (E_b/N_0)_i \\ \text{subject to} & \frac{1}{\mathrm{n}} \sum_{i=1}^{n} BER_i \leq BER_R \\ \text{with} & \mathrm{t}_{\mathrm{n \times m}} \in \{0,1\} \end{array} \qquad \textbf{(2.2)}$$

where BER_R is the maximum allowable average BER value on the communication channel, n is the number of time slot. m is derived from the multiplication of the number of glossary and the number of glossary point through the E_b/N_0 curve. There is one example located in Figure 2.1 to explain the structure. The main goal is to minimize the E_b/N_0 value for the system performance improvement.

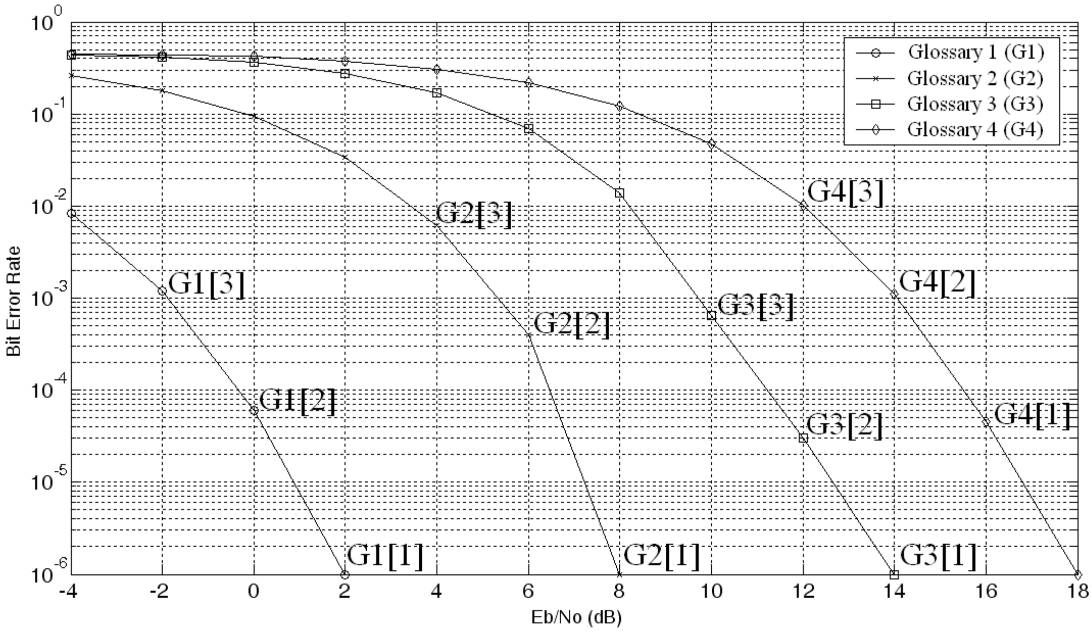

Figure 2.1 : Seleceted points from glossaries in E_b/N_0 space.

The following equation helps to explain the variables in equation 2.2.

$$
\begin{aligned}
&\text{Glossary Matrix}: [G]_{m\times 2} = [G_x]_{mx1} \circ [G_y]_{mx1} \\
&(\mathrm{E_b}/\mathrm{N_0})\,\text{Matrix}: (E_b/N_0) = [t]_{n\times m} \times [G_x]_{mx1} \\
&\text{BER Matrix} \quad : BER = [t]_{n\times m} \times [G_y]_{mx1}
\end{aligned}
\qquad \textbf{(2.3)}
$$

where $[G_x]$ and $[G_y]$ correspond to the E_b/N_0 and BER value of each glossary matrix $[G]$ respectively. Operator "$\circ$" indicates the concatenation of the matrices. When the number of glossary is determined as four and the number of samples for each glossary curve is three as an example like in Figure 2.1. It means that m equals to 12. The number of time slot is defined as 15 (n=15) for this example. Consequently, the size of matrix $[t]$ will be 15x12 and which points to the active glossary with sign "1". This relation between them is given in equation 2.4.

$$
t = \begin{bmatrix}
0&1&0&0&0&0&0&0&0&0&0&0\\
0&0&0&0&0&1&0&0&0&0&0&0\\
1&0&0&0&0&0&0&0&0&0&0&0\\
0&0&0&0&0&0&0&0&1&0&0&0\\
0&0&0&0&0&0&0&0&0&1&0&0\\
0&0&1&0&0&0&0&0&0&0&0&0\\
0&0&0&1&0&0&0&0&0&0&0&0\\
0&0&0&0&0&0&1&0&0&0&0&0\\
0&0&0&0&0&0&0&0&0&0&1&0\\
0&0&0&0&0&0&0&1&0&0&0&0\\
0&0&0&0&0&0&0&1&0&0&0&0\\
0&0&0&0&0&0&0&0&0&0&0&1\\
0&0&0&0&0&0&0&0&0&1&0&0\\
0&0&0&0&0&0&0&0&0&1&0&0\\
0&0&0&0&1&0&0&0&0&0&0&0
\end{bmatrix}
\Rightarrow
\begin{cases}
\mathrm{aG} = \mathrm{G_1}, \mathrm{aP} = 2 \,@\, \mathrm{t} = \mathrm{t_1}\\
\mathrm{aG} = \mathrm{G_2}, \mathrm{aP} = 3 \,@\, \mathrm{t} = \mathrm{t_2}\\
\mathrm{aG} = \mathrm{G_1}, \mathrm{aP} = 1 \,@\, \mathrm{t} = \mathrm{t_3}\\
\mathrm{aG} = \mathrm{G_3}, \mathrm{aP} = 3 \,@\, \mathrm{t} = \mathrm{t_4}\\
\mathrm{aG} = \mathrm{G_4}, \mathrm{aP} = 1 \,@\, \mathrm{t} = \mathrm{t_5}\\
\mathrm{aG} = \mathrm{G_1}, \mathrm{aP} = 3 \,@\, \mathrm{t} = \mathrm{t_6}\\
\mathrm{aG} = \mathrm{G_2}, \mathrm{aP} = 1 \,@\, \mathrm{t} = \mathrm{t_7}\\
\mathrm{aG} = \mathrm{G_3}, \mathrm{aP} = 1 \,@\, \mathrm{t} = \mathrm{t_8}\\
\mathrm{aG} = \mathrm{G_4}, \mathrm{aP} = 2 \,@\, \mathrm{t} = \mathrm{t_9}\\
\mathrm{aG} = \mathrm{G_3}, \mathrm{aP} = 2 \,@\, \mathrm{t} = \mathrm{t_{10}}\\
\mathrm{aG} = \mathrm{G_3}, \mathrm{aP} = 2 \,@\, \mathrm{t} = \mathrm{t_{11}}\\
\mathrm{aG} = \mathrm{G_4}, \mathrm{aP} = 3 \,@\, \mathrm{t} = \mathrm{t_{12}}\\
\mathrm{aG} = \mathrm{G_4}, \mathrm{aP} = 1 \,@\, \mathrm{t} = \mathrm{t_{13}}\\
\mathrm{aG} = \mathrm{G_4}, \mathrm{aP} = 1 \,@\, \mathrm{t} = \mathrm{t_{14}}\\
\mathrm{aG} = \mathrm{G_2}, \mathrm{aP} = 2 \,@\, \mathrm{t} = \mathrm{t_{15}}
\end{cases}
\qquad \textbf{(2.4)}
$$

The aim of PBCS is to solve the nondeterministic problem defined in equation 2.2 and find out the optimum solution for radio link in terms of minimized E_b/N_0 value over the selected time slot. The constraint is the average BER value for the selected time frame. The average of this value should be lower than the acceptable threshold. The penalty function of this optimization is the cost of switching process between the glossaries.

Conventional communication systems utilize some modulation and/or multiple accessing techniques depend on the rules in the standards. Table 2.1 shows the best working points of some selected communication systems.

Table 2.1: The Spectral Efficiency Values for Common Communication Systems.

Service	Standard	Modulation / Multiple Accessing Technique	LSE (bps/Hz)	SNR (dB)
1G	AMPS	TDMA	0.32	2.68
2G	GSM	GMSK / FSK	0.52	12.50
3G	CDMA2000	CDMA / TDMA	2.50	11.61
Wi-Fi	IEEE 802.11 a/g	OFDM / DSSS	2.70	7.50

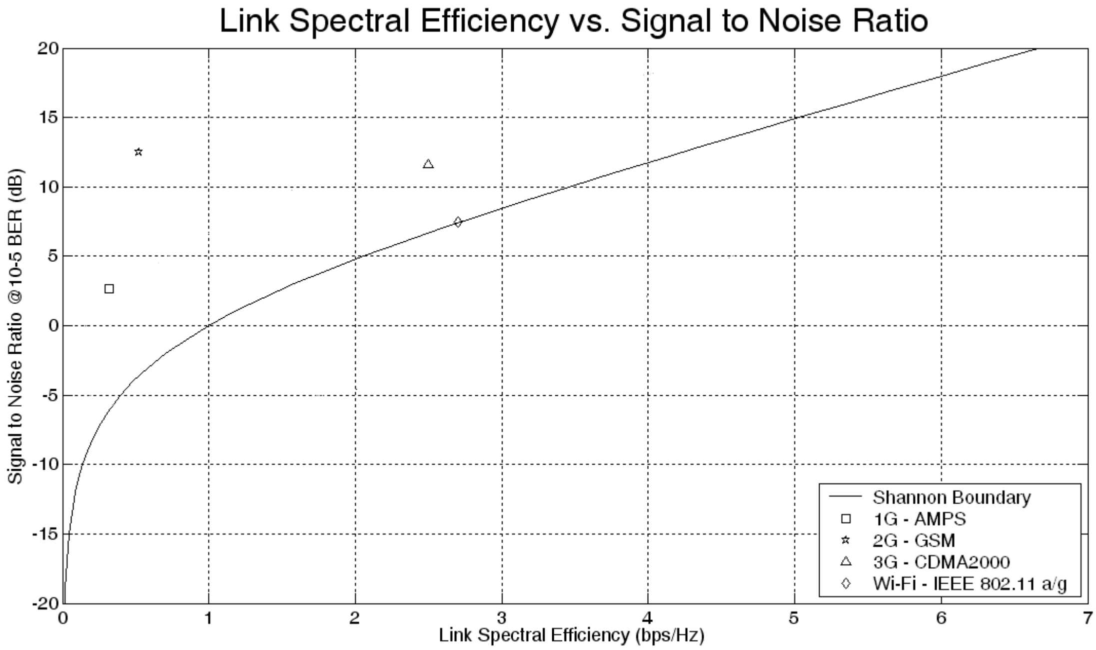

Figure 2.2 : Conventional communication services and PBCS.

1G cellular service was developed in 1981 with NMT-450 (Nordic Mobile Telephone) standard. It is improved by AMPS (Advanced Mobile Phone System) in Bell Labs. Second generation cellular service 2G, was developed with the utilization of GSM (Global System for Mobile Communication) standard. This technology is improved by the usage of CDMA and TDMA. This service is called as 3G. The other popular standard IEEE 802.11a/g has taken the valuable position in wireless communication. This service is named Wi-Fi. It could reach to lower SNR with higher LSE value. Due to fact, this service has the closest performance to Shannon limit in SNR vs. LSE space. The signs in Figure 2.2 show the best performance of the conventional cellular services. On the other hand, PBCS provides more flexible structure instead of the limited flexibility than the conventional cellular services. This feature provides PBCS with an adaptive structure. It will be explained and supported with the simulation results in section 4.

PBCS has manageable SNR capability with the utilization of different glossaries. According to the user expectation, the glossaries can be switched. The system works on the most proper point of the defined space. The adaptive structure of PBC provides conceptual system PBCS with the utilization in software defined or cognitive radio applications.

2.2 PBC and the Related Conventional Modulation and Supporting Techniques

PBC have some features to compare with the conventional methods. This subsection aims to compare PBC with the related conventional modulation and supporting methods in the literature. First, the relation between PBC and well known supporting technique Matched Filter will be evaluated in terms of the robustness against to noise. The most popular adaptive modulation technique QAM will be compared with PBC to evaluate the adaptive performance of the proposed technique. Moreover, Wavelet Modulation (WM) will be discussed next subsection. Then, the most popular modulation technique OFDM in CR applications will be compared with PBCS in terms of pros and cons. Finally, Impulse Radio Ultra Wideband (IR-UWB) applications in CR will be emphasized. The comparison between PBCS and IR-UWB (namely Pulse Shape Modulation - PSM) will be given in detail. Last two comparisons will be in AWGN channel and the results are given in last section. The rest of the comparisons will be used for the future works definition.

2.2.1 Matched Filtering

Matched Filter (MF) is obtained by correlating a known signal with an unknown signal to detect the presence of the template in the unknown signal. The MF is the optimal linear filter for maximizing the SNR in the presence of Additive White Gaussian Noise (AWGN).

A sequence in binary code can be described as a sequence of unit pulses or shifted rectangle functions. Each pulse being weighted by +1 if the bit is "1" and by 0 if the bit is "0". Mathematically description of this generation is shown below,

$$a_k = \begin{cases} 1, & \text{if bit } k \text{ is } 1 \\ 0, & \text{if bit } k \text{ is } 0 \end{cases} \quad \textbf{(2.5)}$$

The binary message can represented by *M(t)*, as the sum of shifted unit pulses:

$$M(t) = \sum_{k=-\infty}^{\infty} a_k \times \Pi\left(\frac{t-kT}{T}\right) \quad \textbf{(2.6)}$$

where *T* is the time length of one bit. An example for this message with ten bits is shown in following figure,

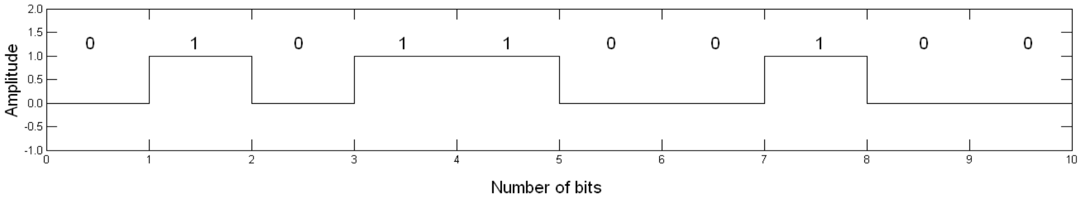

Figure 2.3 : Binary message at transmitter side.

The message at the transmitter is sent across the noisy channel to the receiver. A matched filter can be used to detect the transmitted pulses in the noisy received signal. If the model of noisy channel is determined as AWGN channel, white Gaussian noise is added to the original signal. At the receiver front-end, for a SNR of 3dB, this message can be taken as the following figure.

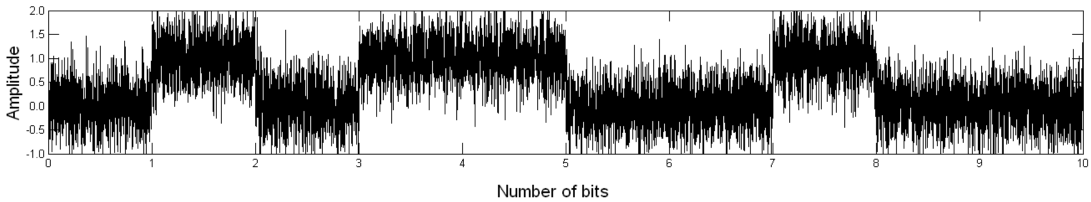

Figure 2.4 : Noisy binary message received by the receiver.

The impulse response of the matched filter convolution system is called *h(t)*

$$h(t) = \Pi\left(\frac{t}{T}\right) \quad \textbf{(2.7)}$$

After convolving with the correct matched filter and received noisy signal, the output signal can be obtained as $M_f(t)$,

$$M_f(t) = M(t) * h(t) \tag{2.8}$$

where "*" denotes convolution operation. The result of this convolution can be show in below figure.

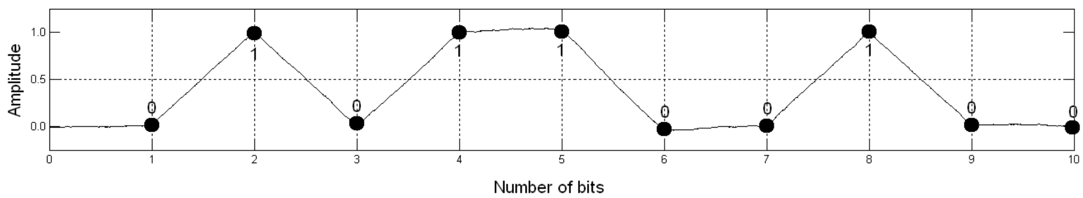

Figure 2.5 : The decoded binary message after the convolution with MF.

The appropriate threshold value provides the exact binary message, which is sent by the transmitter. MF is a proper method to detect the message by extracting the unknown signal with the utilization of known information. This technique is used by the different conventional modulation and encoding techniques to correct the information.

PBC has similar characteristic in terms of detecting the unknown signal by using known pattern. The difference between them is that MF is a supporting method of any modulation and/or encoding technique, although PBC is an adaptive modulation and encoding technique. Moreover, MF needs some additional information at receiver side such as bandwidth, operating frequency, modulation type / order, pulse shaping and frame format etc, but PBC includes all needed information with pre-trained ANN structure at receiver.

2.2.2 Quadrature Amplitude Modulation

Quadrature Amplitude Modulation (QAM) has continued to gain interest for using practical application, since its discovery in the early 1960s. QAM supports both an analog and a digital modulation approaches. It conveys two analog or digital signals by modulating the amplitudes of two carrier waves (Digital case: ASK and Analogue case: AM). This approach is also applied to phase. Two sinusoidal carrier waves with 90° phase shift are used and they called quadrature carriers or components. The modulated waves are summed, and the resulting waveform is a combination of both phase-shift keying (PSK) and amplitude-shift keying (ASK) in the digital case or Phase Modulation (PM) and Amplitude Modulation (AM) in the analog case. A finite number of at least two phases and at least two amplitudes are used in the digital QAM case. QAM’s link spectral efficiencies of 6 bps/Hz can be achieved with 64-QAM. Different level of QAM scheme provides adaptive modulation ability to the structure.

The ideal structure of a QAM transmitter can be explained as follow. First, the bit stream is split into two equal parts to transmit; this process generates two independent signals, they are encoded separately with the utilization of ASK approach. Then one channel is multiplied by a cosine, and the other is multiplied by a the quadrature part, namely sinus. This way creates a phase shift of 90° between them. They are simply added one to the other and sent through the communication channel.

The receiver simply performs the inverse process of the transmitter. The ideal structure is defined as follow. The captured signal can be extracted the component in phase (or in quadrature) by multiplying with a cosine (or a sine). The output should be fed into the a low-pass filter. After this process, the output can be reconstructed by using only an ASK demodulator. At the end, two flows of data can be merged back. In practice, an unknown phase delay between the transmitter and receiver must be compensated by synchronization of the receivers. This technique is well-known in communication world and frequently used in adaptive communication structure.

2.2.3 Wavelet Modulation

Wavelet transforms can be used to generate waveforms that are suitable for transmission over wireless channels in the field of digital wireless communications. This type of modulation is known as Wavelet Modulation (WM) (Manglani, 2001) or fractal modulation. The advantage of this scheme emerges from its diversity strategy: WM allows transmission of the data signal at multiple rates simultaneously. This multi-rate diversity scheme offers advantages in mobile communications. If the channel is not clear for one specific bit rate, meaning that the signal will not be received, the signal can be sent at a different bit rate where SNR is higher. WM is a special kind of multi-carrier modulation signals.

The similarity between WM and PBC is that WM is a special version of Pulse Shape Modulation (PSM), which will be explained below. It uses wavelet transform to modulate the information. Each wavelet function points to pulse shape for signal envelope. PBC also uses a envelope pattern to modulate the information and it is constructed by its innovative algorithm. This envelope is also a pulse shape. The important point is that PBC can produce any kind of pulse shapes to construct an envelope such as wavelet transform function, raised cosine, gaussian etc. This feature provides this new approach more flexible and innovative structure, when PBC is compared with WM.

2.2.4 Orthogonal Frequency Division Multiplexing

Orthogonal frequency-division multiplexing (OFDM) is a well-known frequency-division multiplexing (FDM) scheme. This is a digital multi-carrier modulation method. The main concept to construct OFDM is the frequency spectrum distribution, namely sub-carriers. A large number of closely spaced orthogonal sub-carriers are used to carry data. The data is divided into several parallel data streams or channels, each sub-carrier represents one data stream modulates with a conventional modulation scheme (such as QAM or PSK) at a low symbol rate, maintaining total data rates similar to conventional single-carrier modulation schemes in the same bandwidth.

OFDM has a strong position against to single-carrier schemes. Due to its ability to cope with severe channel conditions (i.e. attenuation of high frequencies in a long copper wire, narrowband interference and multipath effect), OFDM does not need complex equalization filters. OFDM uses many slowly modulated narrowband signals rather than one rapidly modulated wideband signal, hence channel equalization can be simplified. The low data rate makes the use of a guard interval between symbols affordable, making it possible to handle time spreading and eliminate Inter Symbol Interference (ISI). OFDM system model is suitable for a time-invariant AWGN channel. Channel equalization concept and its details will be located in the following sections.

The transmitter block diagram of OFDM can be explained as follow. The serial stream of binary bit $s[n]$ is fed into the system. These are demultiplexed into N parallel streams by inverse multiplexing. Each parallel stream is mapped to a symbol stream using proper modulation scheme (QAM, PSK, etc.).

An Inverse Fast Fourier Transform (IFFT) is computed on each set of symbols. The output gives a set of complex time-domain samples. The real and imaginary components of these samples are converted to the analogue domain using digital-to-analogue converters (DACs); the outputs (analogue signals) are used to modulate cosine and sine waves at the carrier frequency, f_c, respectively. The modulated analogue outputs are quadrature-mixed to pass-band in the standard way. These signals are then summed to give the transmission signal *s(t)*.

The receiver captures the radio signal $r(t)$ with its antenna block. The received signal is quadrature-mixed down to baseband using cosine and sine waves at the carrier frequency. This process creates signals centered on $2f_c$ and low-pass filters are used to reject side bands. The baseband signals are sampled by using analogue-to-digital converters (ADCs). The forward Fast Fourier Transform (FFT) block is used to convert back to the frequency domain. The output of FFT block is N parallel streams, each of which is converted to a binary stream using an appropriate symbol detector. These streams are re-combined into a serial stream $s[n]$, which is the reconstructed signal of binary stream at the transmitter.

The important parameters of OFDM can be listed as

a) *Guard Interval*: It directly relates to symbol duration and Cyclic Prefix (CP). Increment in CP length decreases ISI (Multipath Effect),

b) *Subcarrier Spacing*: It is the distance between utilized communication channels in frequency spectrum. Increment in subcarrier distance decreases Inter Carrier Interference (ICI or Doppler Effect), Doppler Effect is a main problem for OFDM.

OFDM and PBCS can be compared in terms of Cognitive Radio requests to prove the utilization capability of conceptual system PBCS in CR. Due to the scope of this thesis, which is limited by the evaluation only in AWGN channel, the exact evaluation in fading channel for CR will not be given in this thesis. The comparison will be used for the future works.

The CR requests and the responses for these requests of them are listed below;

a) Spectrum Sensing:

   i. *OFDM*: It has inherent FFT block, this causes the reduction of spectrum sensing cost.

ii. *PBCS*: Receiver knows active glossary, it means that if pear-to-pear communication is performed, receiver knows the spectral distribution. Otherwise, PBCS needs to use additional FFT block to sense the spectrum.

b) Efficient Spectrum Utilization & Near Band Immunity (NBI):

i. *OFDM*: It can change the waveform by the turning off some subcarriers. This process provides OFDM with the more robust structure against to NBI.

ii. *PBCS*: The pattern envelope is important for recovering the information; therefore, the central frequency of signal pattern can be adaptively shifted to the free communication channel. This process adds the ability of NBI robustness to PBCS.

c) Adaptation / Scalability:

i. *OFDM*: It can be adapted to different transmission environments by the change on some parameters such as CP size, modulation, coding, subcarrier power.

ii. *PBCS*: It is the same for PBCS, the parameters (the interval for A, F and P) can be changed by its user.

d) Multiple Accessing & Spectral Allocation:

i. *OFDM*: It can perform multiple-access by assigning group of subcarriers to different users (OFDMA).

ii. *PBCS*: Different glossaries can be designed for different users; it provides the communication system with multiple access ability.

e) Interoperability:

i. *OFDM*: It uses different modulation techniques and manages them according to active channel conditions.

ii. *PBCS*: It can not be compared with OFDM in terms of this feature. PBCS has own waveform construction technique and it can only uses them. However, OFDM can use waveform construction technique of

PBCS. This process provides the system with crypto feature for military based application.

Consequently, PBCS has comparable features with OFDM in terms of CR requests for the future evaluation in fading channel. PBCS may need additional cost for FFT block. On the other hand, PBCS is more capable than OFDM to encrypt the information. The other features are depending on the change on some parameters and two of them can satisfy CR requests.

2.2.5 Pulse Shape Modulation in Impulse Radio Ultra Wide Band Systems

Impulse Radio (IR) based Ultra Wideband (UWB) implementation first proposed by Win and Scholtz (1998). Impulse Radio Ultra-Wideband (IR-UWB) is a radio technology that uses pulses to modulate the information. Therefore, it is also called Pulse Shape Modulation (PSM). UWB utilizes very low energy levels for short-range high-bandwidth communications by usage of large portion of the radio spectrum. IR-UWB switches between underlay and overlay mode to perform data communication. It works together with conventional modulation and coding techniques in overlay mode. On the other hand, underlay mode directly equivalent to UWB communication as mentioned above.

A significant difference between traditional radio transmissions and IR-UWB radio transmissions is that traditional systems transmit information by varying the amplitude, frequency, and/or phase of a sinusoidal wave. IR-UWB transmits information by generating radio energy at specific time instants and occupying large bandwidth thus enabling a pulse-position or time-modulation. The information can also be modulated on UWB signal forms (pulses) by encoding the polarity of the pulse, the amplitude of the pulse, and/or by using orthogonal pulses.

IR-UWB offers a variety of options regarding the shapes of the transmitted pulses such as Raised Cosine, Gaussian etc. (Figure 2.6). If the amount of available band decreases, impulse radio elongates the transmitted pulses and uses lower bandwidth. It means that system changes the mode from underlay to overlay. This feature provides impulse radio with flexible structure for CR.

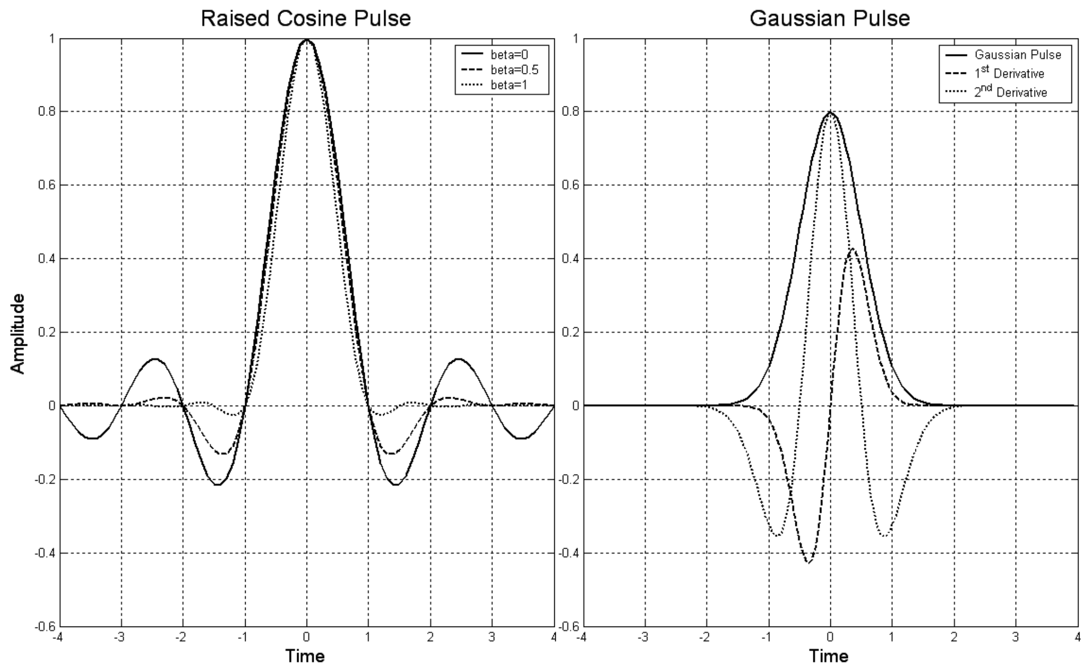

Figure 2.6 : Basic UWB pulse shapes (Raised Cosine & Gaussian).

PBCS utilizes the varying of three features of the signal with sinus form. The pulse shapes (the waveform envelopes) in UWB are the derivative of sinus forms. Therefore, PBCS has a capability to construct same waveform as IR-UWB. PBCS has more flexible and manageable structure than IR with this feature.

The modulation and coding algorithm of PBCS covers IR-UWB and conventional narrow band modulation techniques. PBCS can be related with IR-UWB by the construction algorithm of orthogonal signal forms and the capability of pulse simulation with its structure. However, it does not use different pulse shapes to transmit information as IR-UWB; it utilizes the orthogonal sinus waves with varying amplitude, frequency and phase as conventional shifting and modulation methods. This capability of PBCS provides itself with the same waveform construction capability of IR-UWB by using traditional shifting techniques. Due to fact, PBCS's waveform construction algorithms cover IR-UWB and traditional narrow band modulation techniques.

IR-UWB and PBCS are compared to point pros and cons of PBCS against to IR-UWB in terms of Cognitive Radio requests. This comparison is also performed for the future work due to the same reason, which is mentioned in OFDM section. The CR requests and the responses about these requests of them are listed below;

a) Limited Interference to Licensed Systems:

i. *IR-UWB*: It uses two modes such as “underlay” and “overlay” (Figure 2.11). It provides IR with a significant potential for fulfilling the limited interference.

ii. *PBCS*: The recognition capability is mostly affected by the frequency and phase changes. PBCS can work on underlay and overlay modes easily. LPEC and SPEC algorithms correspond to underlay and overlay modes respectively. These algorithms will be defined in section 3.

b) Dynamic Spectrum Utilization:

i. *IR-UWB*: Varying the duration of the pulses form directly alters the occupied spectrum.

ii. *PBCS*: Varying the frequency and phase interval directly alters the occupied spectrum.

c) Dynamically Adjustable Data Rate:

i. *IR-UWB*: Duty cycle determines the data rate and it can be adjusted adaptively.

ii. *PBCS*: The bit level of glossary determines the data rate and it can be adjusted adaptively.

d) Adaptable Transmit Power:

i. *IR-UWB*: The amplitude of pulse shape directly determines the transmitted power. It can be changed easily to provide adjustable transmission power.

ii. *PBCS*: The boundary of amplitude for envelope directly determines the transmitted power. It can be changed easily to provide adjustable transmission power.

e) Adaptive Multiple Access:

i. *IR-UWB*: Each user utilizes different values for number of chips in a frame, duty cycle and transmits power. It provides IR with multi-user support.

ii. *PBCS*: Each user utilizes different glossaries. It provides PBCS with multi-user support.

f) Information Security:

i. *IR-UWB*: Underlay mode has low transmit power so the licensed user could not listen this information. Overlay mode uses its own communication band.

ii. *PBCS*: It has capability to switch to underlay and overlay modes but additionally it has naturally crypto feature by the utilization of different glossaries. The user has this information because of pre-trained structure.

g) Limited Cost:

i. *IR-UWB*: It depends on pulse shaping so the basic UWB RF-Front-end can satisfy the radio layer with low cost.

ii. *PBCS*: The resolution of the pattern is important and affects the recognition performance. It needs to use Direct RF Sampling in Radio Layer. This is not low cost solution.

Consequently, PBC has strong noise robustness and so it is compared with the well-known supporting technique, namely Matched Filter. Moreover, it is compared with QAM in terms of adaptive capability in AWGN channel. These two evaluations defines the scope of this thesis. On the other hand, it is compared with OFDM and PSM (including the special version of WM) in terms of the CR application capability. However, this comparison is only performed conceptually, and shared the comments about these evaluations. According to them, first comment is that each glossary in its glossary space has pre-defined spectral distribution and it can be managed by OFDM structure. It means that PBC and OFDM can be used in the same communication system. This flexible structure provides PBC with the utilization of adaptive communication structure as CR. Second, PBC uses basic shifting rules to construct waveform. Therefore, it simulates all conventional shifting based modulation techniques. In addition to this, the waveform construction algorithms have a capability to simulate pulse signal as IR-UWB. This feature provides PBCS with the more flexible structure for the utilization of adaptive modulation based applications.

2.3 Design Considerations of PBCS in Fading Channel

There are two main design restrictions for PBC in high frequency RF communication. First, it needs high sampling resolution to increase the recognition performance of ANN. Direct RF sampling can be a solution for this need. Second restriction is coming from the delay and reflections in the communication medium. This is named fading problem. Although this study mainly focuses on the Additive White Gaussian Noise (AWGN) channel model for baseband application in Ultra Low Frequency (ULF) communication band, the Multipath and Doppler effects in radio link are not ignored and defined as future works. Therefore, some studies in the literature related to these problems are investigated. Some proper methods are selected and explained in this thesis. Following subsection contents these design restrictions and solution advices for them.

2.3.1 Radio Block - Direct RF sampling

PBCS depends on the pattern recognition. The sampling resolution of the signal pattern directly affects the performance of the recognition layer. The sampling frequency of data converter (Analog to Digital Converter (ADC) and/or Digital to Analog Converter (DAC)) is limited by the integrated circuit technology. Direct RF sampling (Akos et al., 1999, Akos, 2004) is a radio receiver technique. It provides its user with the sampling frequency that is much lower than the original carrier frequency. Direct RF sampling maps the communication channel to the baseband. According to Nyquist theorem, minimum sampling frequency is selected two times of only the spectral bandwidth for the related communication channel.

RF front-end used direct RF sampling technique simultaneously receives multiple frequency bands. This extended version of Direct RF sampling approach was proposed by Psiaki et al. in 2005. This block includes a Low-Noise Amplifier (LNA), a Multiband Band-Pass Filter (MBPF) and an ADC with a sampler. The MBPF passes several frequency bands of request and rejects all others. According to this approach, the frequency bands of interest are centered at the three widely separated carrier frequencies. If the ADC sampling frequency is f_s, it maps each of the frequency bands to non-overlapping portions of the Nyquist bandwidth from 0 to $f_{s/2}$. The aliased versions of the original carrier frequencies are relocated to the frequency response space in between 0 and $f_s/2$. As a result, the sampling frequency for signal patterns directly related to the utilized spectral bandwidth. If PBCS uses direct RF sampling block at front-end, the hardware restriction for sampling resolution disappear.

2.3.2 Channel Estimation Techniques

Channel estimation is the complementary part of the adaptive receiver design for narrowband communication. The avoidance from frequency selectivity (Multipath Effect) and time selectivity (Doppler Effect) are two prominent issues for communication system. The proposed system should solve these problems with its features or the conventional approaches. This section aims to point the solution for these effects in PBCS, when the performance of it will be evaluated in fading channel.

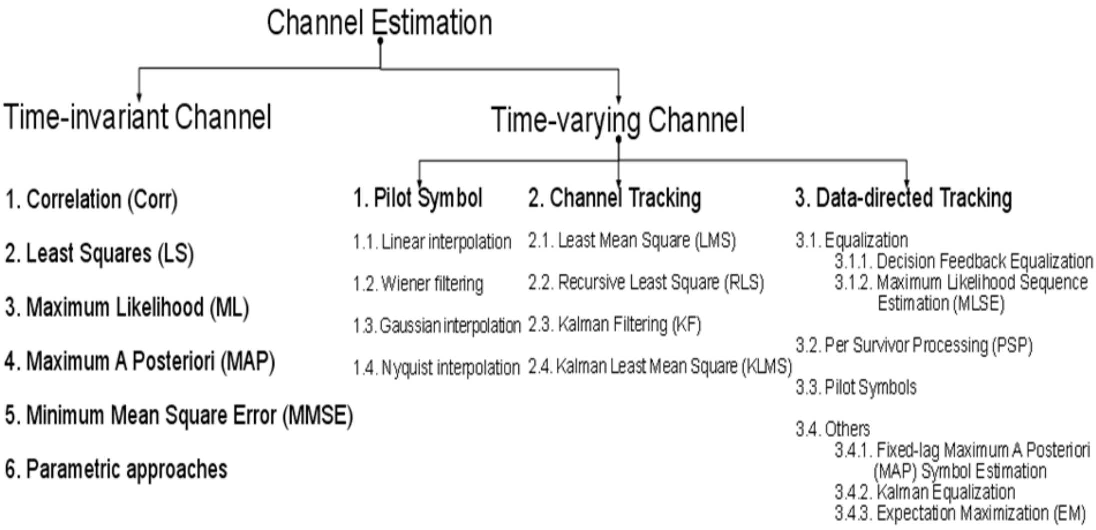

Figure 2.7 : Channel estimation methods.

Two main system models are considered for channel estimation. These channel models are the time-invariant and the time varying channels. Time-invariant channel model faces with frequency selectivity effect. Time varying channel model is more complex than the time-invariant and it causes time selectivity problem. This effect generally occurs in mobile communication systems due to the movement at the receiver side. The chart for channel estimation methods are given in Figure 2.7. This chart is derived from the survey published by Arslan and Bottomley (2001).

The mathematical expression of channel models for time-invariant channels can be given as

$$\vec{r} = \vec{b}\,\vec{h} + \vec{z} \tag{2.9}$$

where $\vec{r}$, $\vec{b}$, $\vec{h}$ and $\vec{z}$ corresponds to received signal, transmitted signal, channel coefficients and noise models respectively. The assumption for calculating channel coefficients is that the receiver always knows the training sequence constructed by the original symbols. This assumption overlaps PBCS structure. In PBCS, the receiver has pre-trained data set by the symbols at transmitter side. Therefore, all of the mentioned list below time-invariant channel in Figure 2.7 can be used in PBCS. The differences between each approach is explained below.

1. Correlation (CORR): Transmitter and receiver know the data set. The received signal is shifted and each output is used for calculating the correlation coefficients between the known sequences.

$$\hat{h}_{CORR}(k) = \frac{1}{N_S} \sum_{l=l_0}^{l_0+N_s-1} r_{k+l} b_l^{*} \tag{2.10}$$

where, k, l and N_s corresponds to the index of channel coefficient, the index of tap and the utilized number of symbols for all situation. "*" in this formula denotes complex conjugation.

2. Least Squares (LS): It calculates the minimum difference between known and received signal with least squared approach. This is more suitable than correlation for multi-tap situation.

$$\hat{\vec{h}}_{LS} = \left(\vec{b}^{H} \vec{b} \right)^{-1} \vec{b}^{H} \vec{r} \tag{2.11}$$

where, H denotes "Hermitian" operator for the vector. It means that the complex conjugate of its transpose for any matrix.

3. Maximum Likelihood (ML): The difference between LS is the noise model. The noise covariance matrix should be added to the formulization. R_Z denotes this matrix in equation 2.12.

$$\hat{\vec{h}}_{ML} = \left(\vec{b}^H \, \vec{R}_Z^{-1} \, \vec{b} \right)^{-1} \left(\vec{b}^H \, \vec{R}_Z^{-1} \right) \vec{r} \tag{2.12}$$

4. Maximum a Posteriori (MAP): The channel priori information is used at this calculation. The formula includes channel coefficient covariance matrix. R_h corresponds this matrix in equation 2.13.

$$\hat{\vec{h}}_{MAP} = \vec{R}_h \, \vec{b}^H \left(\vec{b} \, \vec{R}_h \, \vec{b}^H + R_Z \right)^{-1} \vec{r} \tag{2.13}$$

5. Minimum Mean Squared Error (MMSE): This estimation uses the channel coefficients as random variable. MMSE is equivalent to the MAP under the Gaussian channel coefficients assumption.

6. Parametric Approaches: The pulse shape auto-correlation matrix $\vec{a}$ is defined for these approaches. The form of $\vec{b}$ goes to $\vec{ba}$ and the calculation performs with this approach.

Time varying channel models are divided in three main parts. These are shown in Figure 2.7. PBCS in this thesis is designed for time-invariant channel model. However, this design can be expanded with time varying channel model. In this case, intelligent layer approach will be the best fit to this model. Online training of ANN with received sequence or Genetic Algorithm utilization for channel estimation overlaps with PBCS approach. This approach is similar to Per Survivor Processing (PSP) in the literature. The following sections will be evaluate and explain the Multipath and Doppler effects with the solution advices.

2.3.2.1 Frequency Selectivity (Multipath Effect)

Multipath time delay estimation (MTDE) is an important problem in the field of RF communication. Many studies in the literature consider the problem of estimating the arrival times of overlapping ocean-acoustic signals from a noisy received waveform.

The estimating problem consists of attenuated and delayed replicas of a known transient signal. Many algorithms in the literature are proposed to solve this kind of problems. These algorithms can be evaluated in techniques pointed in Figure 2.13. These selected techniques listed as expectation maximization algorithm (Feder and Robinson, 1988), regression stepwise procedure (Li, 1998), L-1 norm regularization algorithm (Fuchs, 1999) and MODE-WRELAX algorithm (Wu et al., 1999), etc. Other studies assume that the received multipath signal is a summation of multiple time-delayed replicas of a continuously random process, the maximum-likelihood method and autocorrelation estimator based algorithms belong to this category.

One of the proper methods (Ebrahimi and Tabatabavakili, 2007) for the proposed system uses Genetic Algorithms (GA). It determines the arrival time of multipath and the attenuation value of communication signal. This method assumes that the receiver knows the transmitted signal and the number of paths in the multipath environment. Another assumption is that the received signal from front end is distorted in the communication medium with only Additive White Gaussian Noise (AWGN). The performance of GA is examined for different signal-noise scenarios in that study. The simulation results show that the time delays are estimated well based on the GA. This technique determines the objective function as the minimizing error between expected signal and received signal in terms of Mean Squared Error (MSE). This algorithm can find the global optima on the time space. This point refers to the beginning point of the multipath in time domain. This approach directly uses channel estimation method MMSE in Figure 2.7.

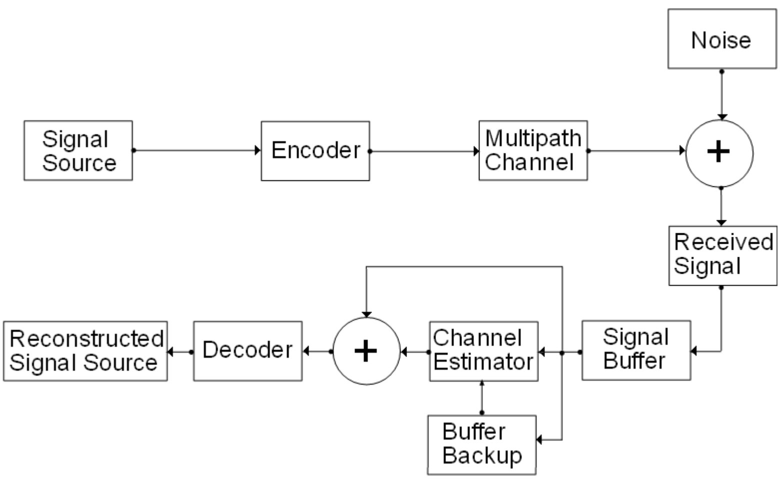

Figure 2.8 : Communication block diagram with multipath & noise model.

ANN is another solution to this problem. The simulation results in this study show that ANN can separate the combined signal. He and Li (1991) have already proven this ability of ANN with their study.

Figure 2.8 shows the common communication block diagram with multipath and noise model. According to this figure, the “Signal Buffer” block takes first part of signal. If it is assumed that the initialization time of the communication does not include multipath effect, this part only includes noise model. Therefore, PBCS can recover this signal pattern easily. This known part is stored in the “Buffer Backup” block to determine the multipath arriving time. “Channel Estimator” defined in study (Ebrahimi and Tabatabavakili, 2007) uses this backup and the buffered real time signal pattern. It gives the output such as the arrival time of multipath and the attenuation rate. If PBCS receiver knows the attenuation rate and arrival time of multipath, the system can be simplified and represented as only AWGN model.

Figure 2.9 : The solution advice for multipath model in PBCS.

However, the reflection and time delay always occur in a communication channel and the effect of them cannot be ignored. Time delay and attenuation rate in amplitude can be estimated by the utilization of ANN features. An approach to solve this problem is proposed in this thesis. According to this proposal, the space pattern is constructed by the constant amplitude, frequency and phase values. The time frame for this pattern (t_{space}) is adjusted to the same value for the time frame of pattern (t_{data}). This space pattern is concatenated to each pattern in glossary. ANN is trained by this concatenated pattern. If the time delay (ΔT) does not exceed the total frame time of glossary pattern ($T_{pattern}$), ANN can recover the information. This situation is given in Figure 2.9.

ANN has high recognition capability for the phase and frequency changes at any pattern. The proposed approach benefits from this feature of ANN. If the trained patterns are constructed by the utilization of two subparts such as the variable and constant values of some features (such as frequency, phase), ANN can recover the information by the determination of changes at these features. The important parameter to increase the performance is that the total delay in the last tap should be lower than the duration of the pattern. The simulation results for this proposal will be given in the section 4. Frequency selectivity problem can be solved by above approach or using any conventional channel estimation model pointed out in Figure 2.7.

2.3.2.2 Time Selectivity (Doppler Effect)

PBCS is not designed for time varying channel model in this thesis. However, some different approaches and the solution advices will be located at this section for time selectivity. Doppler effect occurs, if the symbol pattern frame could not be determined properly. Therefore, if the received signal can be parsed in appropriate way and the boundary of the pattern frame can be determined properly, the sampling frequency can be adjusted with the related parameters such as duration of signal pattern and the number of sample for each signal pattern. This situation does not affect the recognition performance of ANN. Adaptive low-pass filter of Angle-of-Arrival (AoA) approaches are the proper ways to determine right frame.

The other solution advice for Doppler Effect overlaps with Per Survivor Processing (PSP). According to this approach, Raheli et al. (1995) use a kind of basic genetic algorithm approach. The pruning process that used for all data-directed tracking methods is performed as the protection of valid path for the solution. This approach is proper to use in PBCS structure with intelligent layer utilization.

The mobile communication systems and the performance evaluation of PBCS under Doppler Effect can be defined as future work. This topic will not be covered in this thesis.

2.4 PBCS in Wireless Communication

Some standards/protocols are defined in the communication world to improve and standardize the communication systems. Quality of Service (QoS) is one of the most important concepts. QoS has the ability to provide different priority to different applications, users, data flows or to guarantee a certain level of performance to a data flow. A network or protocol that supports QoS may agree on a traffic contract with the application software and reserve capacity in the network nodes. It may monitor the achieved level of performance, for example the data rate, delay and dynamically control scheduling priorities in the network nodes. It may release the reserved capacity during a tear down phase.

Another important concept is Wireless Application Protocol (WAP) in wireless communication world. WAP is an open international standard for application-layer network communications in a wireless-communication environment.

Two of them uses Open System Interconnection (OSI) model as a reference. OSI is an abstract description for layered communications and protocol design of computer network. It divides network architecture into seven layers. These are the Application, Presentation, Session, Transport, Network, Data Link, and Physical Layers (Table 2.2).

Table 2.2: OSI Reference Model.

OSI Reference Model			
	Data Unit	Layer	Function
Host Layers	Data	7. Application	Network process to application
		6. Presentation	Data representation, encryption and decryption
		5. Session	Interhost communication
	Segments	4. Transport	End-to-end connections and reliability, flow control
Media Layers	Packet	3. Network	Path determination and logical addressing
	Frame	2. Data Link	Physical addressing
	Bit	1. Physical	Media, signal and binary transmission

Communication systems utilize different modulation techniques to form the source information. Each modulation technique has different performance capability in terms of LSE and the robustness against to noise suppression. CR concept in communication world aims to improve the overall performance of communication systems. It switches the modulation techniques and tries to find the best performance in the specified communication channel. Many different classification and/or clustering methods are used to sense the communication channel availability for this purpose. Neural network usage is one of the most popular methods to sense and determine the free space in the communication channel. The output of neural networks guides CR systems to select the most proper modulation technique for the related communication channel. The usage of ANN at these structures covers the second layer of OSI (Data link layer).

On the other hand, the novelty point of conceptual system PBCS is the utilization of ANN in Physical Layer of OSI reference model. It recognizes its own communication signal patterns in pre-defined glossaries and synchronizes the information at distinct layers. The fundamental reason for selecting ANN as the recognition layer of the system is the sensitivity of ANN while recognizing the amplitude, frequency and phase changes in the communication signal. This feature also affects the synchronization performance of the system.

The control unit, which will be introduced in the next section in detail, of PBCS can be located on Data Link Layer. As mentioned in first section in Figure 1.6, it has not a cognitive engine capabilities with this structure but it can be extended to cognitive engine to design CR with PBC modulation technique.

The next section contents the architecture of PBCS. The transmitter structure will be given with the waveform construction algorithms. The receiver architecture will be explained with ANN design.

3. PBCS ARCHITECTURE & STRUCTURE

Communication systems consist of two main parts as transmitter and receiver. The proposed communication system depends on pattern based encoding in transmitter and artificial neural network based decoding in receiver. The architecture of this system and the structures for sub-parts are given in this section.

3.1 PBCS Architecture

PBCS is designed for the multi-purpose utilization such as the single-user and multi-user access. Different information sources (S_x, x = 1…n, in Figure 3.1) can be located in the transmitter side. While the digital information is encoding, each binary word is assigned to the artificially constructed patterns, namely waveforms. Due to this procedure realizes the modulation of the signal, there is no need for additional modulation in transmission. These modulated signals can be simultaneously transmitted through communication medium. The natural noise is automatically added to these different signal patterns on the communication medium. The antenna at the receiver side percepts the combined and distorted information. Recognition layer can decode this received and synchronized signal. Figure 3.1 shows the generalized view of PBCS structure.

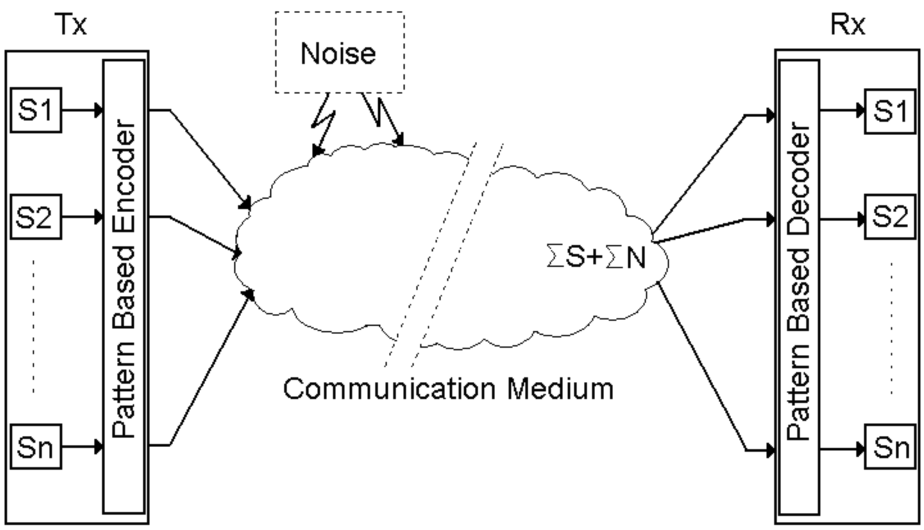

Figure 3.1 : Pattern based communication system (PBCS) structure.

The general description for the proposed system can be explained systematically. The pattern encoder captures a group of bits in the sequence (information source). It directly encodes them into a continuous signal pattern in time domain. Since the waveform is designed to fit in the communication channel, additional modulator is not needed. Receiver directly filters the baseband with RF front-end. The amplified signal is given as an input to the pattern recognition layer. Recognition layer determines the pattern and identifies the binary response of this analogue signal. Bit sequencer concatenates this binary information in an order. Hence, the original information is recovered by using the pattern glossary mutually known by the transmitter and receiver sites.

3.2 Transmitter Structure

The sequence of information is fed into a buffer. According to the size of the glossary, buffer takes the n-bit sequence from this information. This n-bit binary sequence is matched with any n-bit glossary (i.e. the binary sequence "010" is mapped to second pattern in selected 3-bit glossary). The encoder output is fed into the pattern sequencer. It concatenates these patterns in sequence. The concatenated pattern sequence is transmitted through antenna (in special case, RF) Front-end. As seen, by not requiring any modulation and/or coding layer, glossary construction algorithm (PBC modulation and coding technique) forms the communication signals. The left hand-side of Figure 3.2 shows the transmitter structure of PBCS.

"Glossary Selector" block has the most important role for PBCS performance. This block takes the glossary space information and channel spectral situation as an input. According to these inputs, this block calculates the maximum likelihood value. It uses hysteresis function to determine the set of most proper glossaries in the glossary space. The reason of the hysteresis approach utilization in the glossary selector block is that the system gives the priority to the active glossary to reduce the communication cost. The encoder uses the information of the bit stream size at the buffer and it selects the best-fit glossary from the glossary set given by the glossary selector.

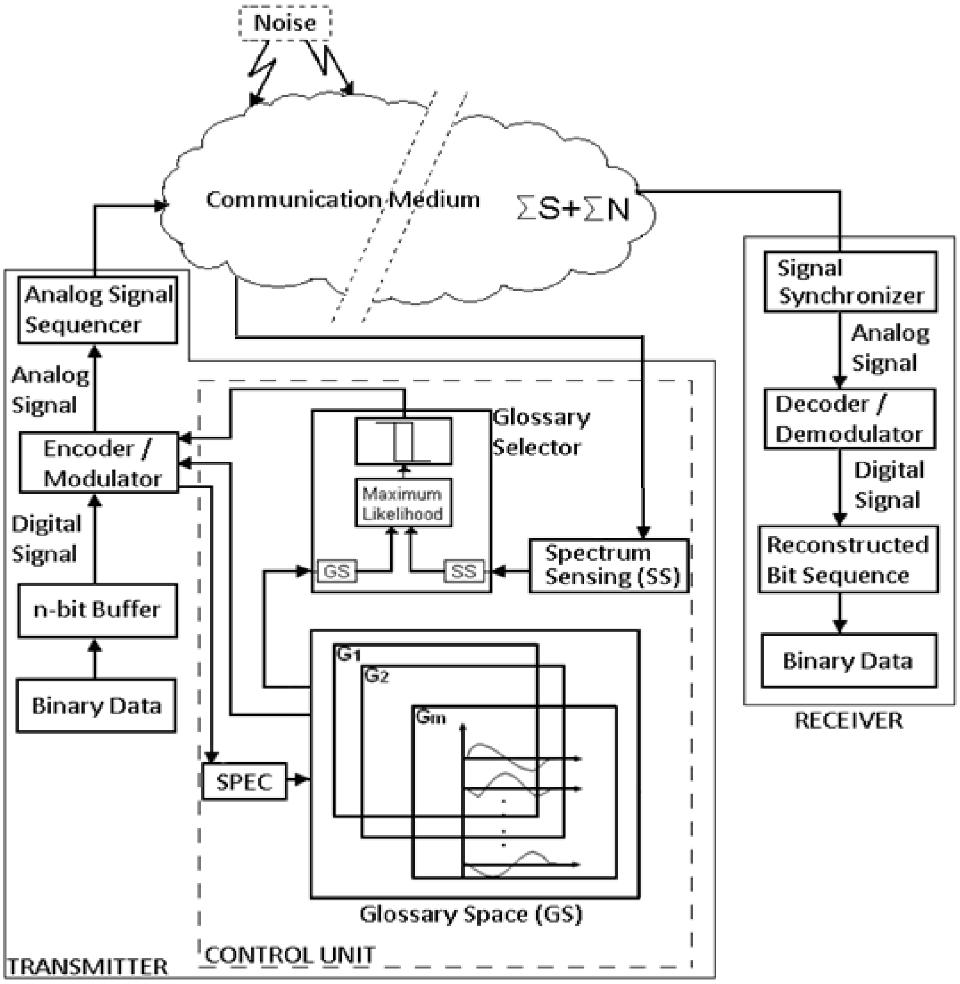

Figure 3.2 : PBCS transciever.

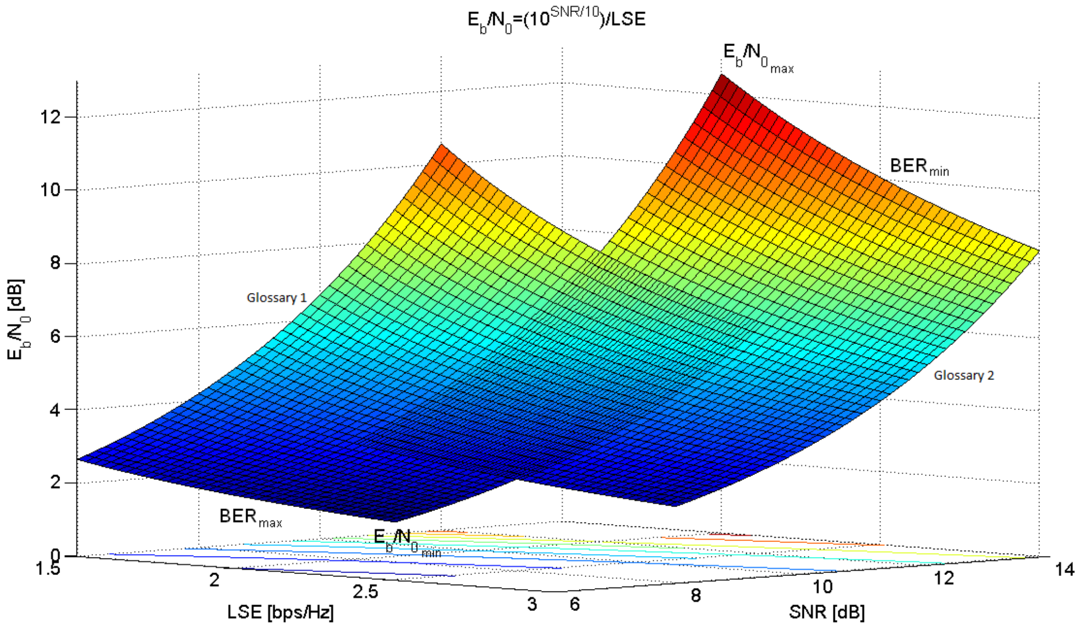

Figure 3.3 : Glossary Selection.

Figure 3.3 represents one example to explain the hysteresis approach for glossary switching process. If the system works with Glossary 1, it does not change the glossary until the valid plane ends. Due to the objective function is to minimize E_b/N_0 under BER restriction, Glossary 1 always works at better conditions then Glossary 2. When the channel condition changes, Glossary 2 should be activated by the system. For the reversing approach, the switching from Glossary 2 to Glossary 1 should not be performed until Glossary 2 is not satisfying the optimization conditions to prevent the additional switching cost. This approach called as hysteresis approach.

3.2.1 New adaptive modulation approach

The glossary construction (modulation and coding algorithm) is one of two innovative parts in PBCS. The aim of this study is to construct the most robust signal patterns against to noise. Due to fact, different glossary construction algorithms are proposed and these are compared to find out the best proper technique. Since the pattern recognition layer is chosen to be Artificial Neural Network (ANN), a signal pattern construction algorithm should be developed to optimize the recognition capability of this layer. Amplitude (A), Frequency (F) and Phase (P) changes are taken in to consideration during the construction of signal patterns. This modulation technique is called Pattern Based Communication (PBC) based on Amplitude Frequency Phase Shift Keying (AFPSK).

There are some prerequisites to construct signal patterns (waveform) in the glossary. In terms of a fundamental necessity of the RF communication, the constructed signal patterns should satisfy the following three conditions:

1) *Zero DC component*: Average of each signal pattern and the series of patterns must be zero, so that the RF energy can efficiently be transferred.

2) *Continuity*: First and second derivatives of the signal patterns and the consequent patterns must be continues in order to prevent non-transferrable harmonics beyond the pass-band.

3) *Bandwidth limitation*: Spectral distribution of the generated signal should ideally be inside the pre-defined frequency interval.

Two main methods are developed for constructing the glossary. These are called linear and sinusoidal pattern envelope construction techniques.

3.2.1.1 Glossary producing with linear pattern envelope construction algorithms

Three different algorithms are developed based on the phase changes in Linear Pattern Envelope Construction (LPEC). These algorithms are labeled as Rotational, Sequential and Interval. The changes in amplitude and frequency in the first two algorithms are shown in the formula below.

$$a_1 = A_{\min}$$
$$a_{k+1} = a_k + \left(A_{\max} - A_{\min} / (\mathrm{SP} - 1)\right) \tag{3.1}$$

$$f_1 = F_{\min}$$
$$f_{k+1} = f_k + \left(F_{\max} - F_{\min} / (\mathrm{SP} - 1)\right) \tag{3.2}$$

The boundary of A and F are defined by the user's entry of a minimum and a maximum value for each signal component. The linear change is applied in each interval. The difference between the maximum and the minimum values of A and F is divided to the intervals by the number of sub-patterns (SP). The third algorithm is named Interval method, the A is kept as in equation 3.1, while the F value is shifted as in formula below.

$$f_1 = F_C - \Delta F$$
$$f_{k+1} = f_k + \left(2 \times \Delta F / (\mathrm{SP} - 1)\right) \tag{3.3}$$

In LPEC algorithms, the amplitude and frequency changes are performed as explained above. However, the name of the algorithms comes from the phase changing method.

In the "Rotational" algorithm, the phase change is made to start from zero and to return to the initial point in each sub-pattern. In the example at Figure 3.4, the phase change is set 360 degrees within 4 sub-patterns. In this process, the stopping points of the linear phase changes are assigned to 90, 180, 270 and 360 degrees. These points can be used in sequence or randomly. The signal pattern in any glossary is produced with the phase shifting from zero to the related degree and back to the initial value for each sub-pattern. Four sub-patterns construct one glossary member.

If the user enters the value for phase as 300 degrees, these interval values automatically is set to 75, 150, 225 and 300 degrees.

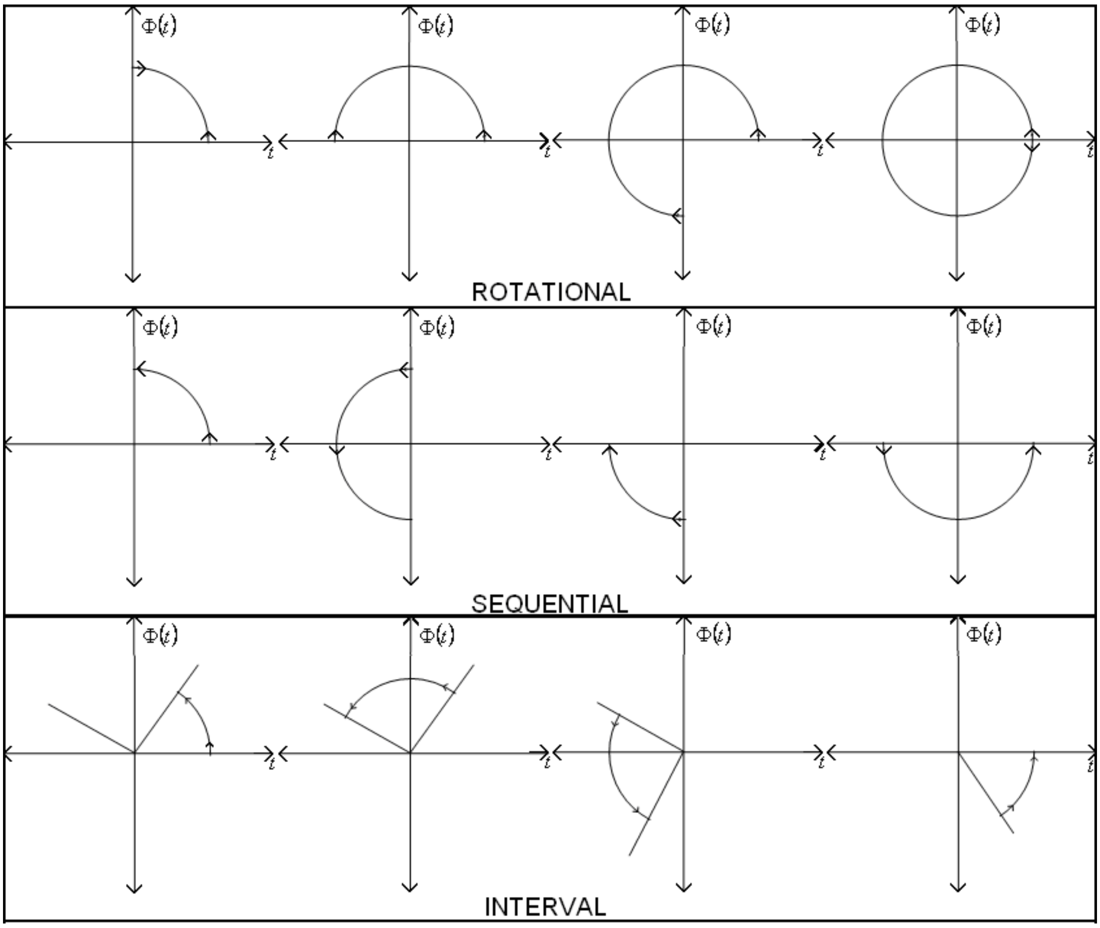

Figure 3.4 : Phase shifting algorithms for LPEC.

In the “Sequential” algorithm, the phase stops at the pre-determined values and finally returns to the initial point. According to the same figure, phase shifting starts with zero degrees, and then stops at 90, 270 and 180 respectively, finally it backs to zero. The intermediary points are selected randomly. Similar to above example, the user determines the phase range. If the user selects the phase limit as 200 degrees, the intermediary points are 50, 100 and 150 degrees. Due to a smoother phase shifting, the “Sequential” algorithm provides less spectral bandwidth utilization than the “Rotational” algorithm.

In the Interval algorithm, the number of the randomly selected intermediary points equals to the number of sub-patterns. The initial value for phase is set to zero. It moves to randomly selected value. This end point equals to the initial value for the next sub-pattern. As the other algorithms, the user determines the phase range but the intermediary points are determined randomly. This process provides an even more effective utilization of the spectral bandwidth. Figure 3.4 is drawn for the user input for phase range as 360 degrees.

Three different pattern construction algorithms are proposed above. According to the spectral analysis, the best fit to the proposed communication system is the Interval method. This is improved and labeled as “Enhanced Interval” algorithm after a minor change in amplitude. The same process for frequency in “Interval” method is applied to the amplitude for performing “Enhanced Interval” algorithm. However, these waveform construction methods need wideband utilization in frequency spectrum space. PBCS can work as underlay mode with waveform produced by LPEC algorithms or PBCS can utilize from band-pass filter to work in narrow band communication.

Since all of these algorithms can produce infinitely many signal patterns, this quantity should be constrained based on the bit level of the glossary (i.e. 4-bit glossary contains 16 signal patterns). This constraint is handled by the well-known clustering method, namely Vector Quantization (VQ), Linde et al. (1980) first proposed this method. The distance parameter of the quantization procedure is chosen as the correlation factor between the signal patterns.

3.2.1.2 Glossary producing with sinusoidal pattern envelope construction algorithms

All of the proposed algorithms above deal with the robustness to noise and proper to use at underlay mode without filtering process. On the other hand, the spectral bandwidth utilization is another important parameter that awaits further concern for narrowband or spread spectrum communication. According to PBC approach, three fundamental features of the communication signal need to be shifted by AFPSK method in order to construct the glossary space. Since the efficient use of spectral bandwidth is concerned, this shifting operation needs to be sinusoidal instead of linear for the utilization of narrowband communication without band-pass filter. The quarter sinus form is used as the basis units in the shift keying. This choice prevents the unexpected extra spectral usage. As a fundamental necessity of the RF communication, the signals pattern should end at its initial point in order to have a zero power density in average. This feature provides a perfect fit during the pattern synchronization. The proposed alternative algorithm is named Sinusoidal Pattern Envelope Construction (SPEC).

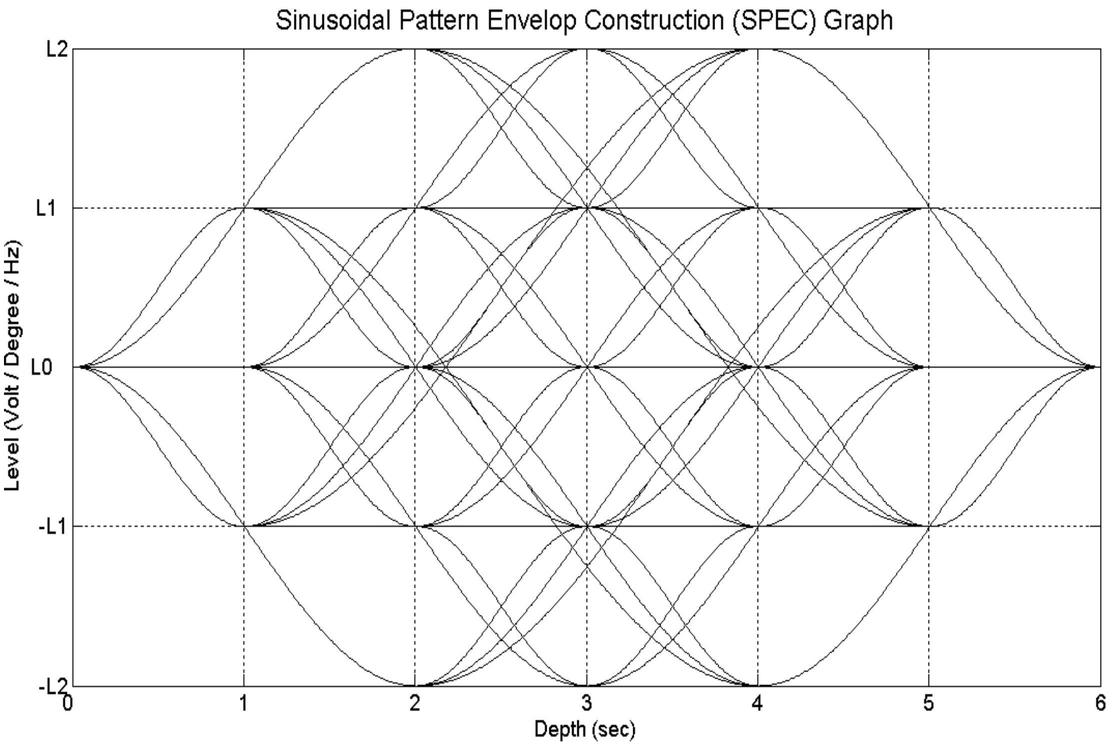

Figure 3.5 : SPEC space for Level: ±2 and Depth:6.

Figure 3.5 shows the specified SPEC space for the envelope combinations of ±2 levels and 6 depth layers. The vertical axis (Level) in the SPEC space represents the maximum and the minimum values of any signal characteristics (amplitude, frequency or phase). For instance, in a case of phase change equals to 180 degrees, the space is defined between the limits +L2 = +180 and –L2 = -180; shifting is made between these two values. Similarly, when the frequency range is chosen between 985 Hz and 1015 Hz, these are assigned to –L2 and +L2 levels respectively. On the other hand, the horizontal axis (Depth) in the SPEC space shows the signal pattern length in terms of the time. The Depth layer provides more alternative routes on the SPEC space but it does not create more inter symbol interference. Because, it only defines the shift keying layers and does not affect the physical length of the communication signal.

All possible outcomes in the SPEC space are formed by the changes in A, F and P features of the signal. There are three possible alterations between each depth layer. These are positive or negative direction on the y-axis (Level) defined by the half sinus function or constant value between the pre-defined levels. The amount of chosen signal envelops is defined as the modulation level of the communication system in terms of the number of bits. The glossary should contain the most uncorrelated signal envelopes. Therefore, SPEC space is used in different ways for the changes of A, F and P. The strategy for A and F is shown in same figure (Figure 3.6). For instance, the bold lines in this figure represent two different envelope options for A or F, these lines show two uncorrelated routes in this defined space. For this example, two routes follow the opposite ways to construct uncorrelated signal forms in terms of A or F. If F is taken into the evaluation as an example, when the envelope route goes down, the frequency of carrier signal decreases. On the other hand, it increases for the opposite direction. The similar scenario for P can be shown as Figure 3.7.

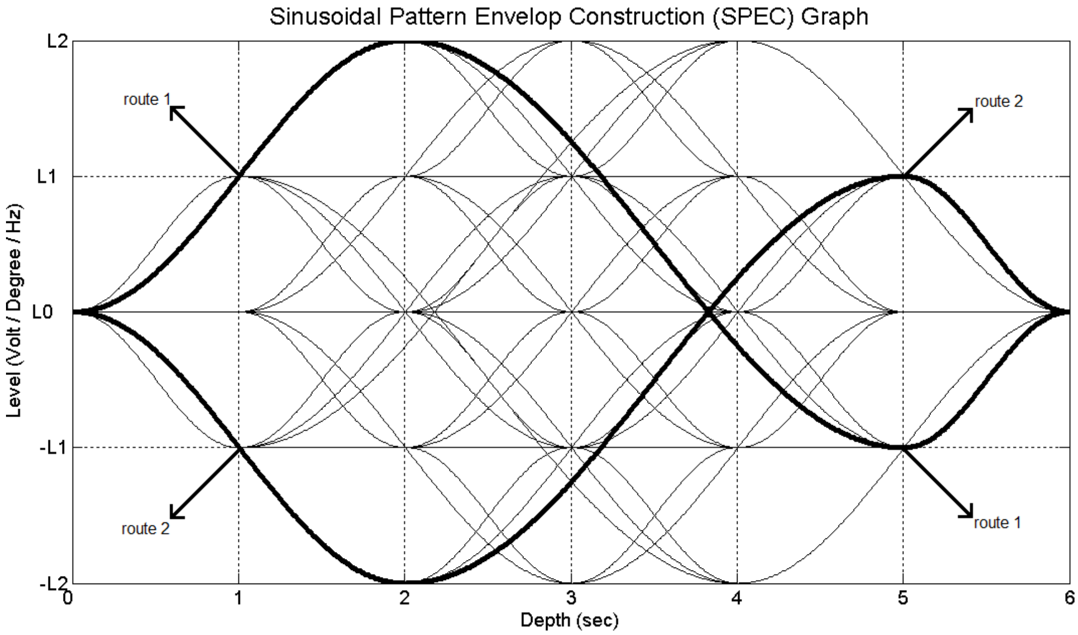

Figure 3.6 : One alternative route for A and F in SPEC space.

The phase difference between two signals should be 90^o to satisfy the orthogonality condition. Since the initial point of any alternative route should be at zero, satisfying the above condition is not possible from the beginning of the communication signal to the end practically but the aim of the algorithm is to find the most uncorrelated alternatives. Therefore, the proposed algorithm uses pair routes and one follows the other as much as 90^o phase shift. Figure 3.7 shows one of the possible routes for this scenario. Orthogonal form can be observed between depth 1 and depth 5. The mathematical expression for the envelope construction of A is given in equation 3.4 as an example.

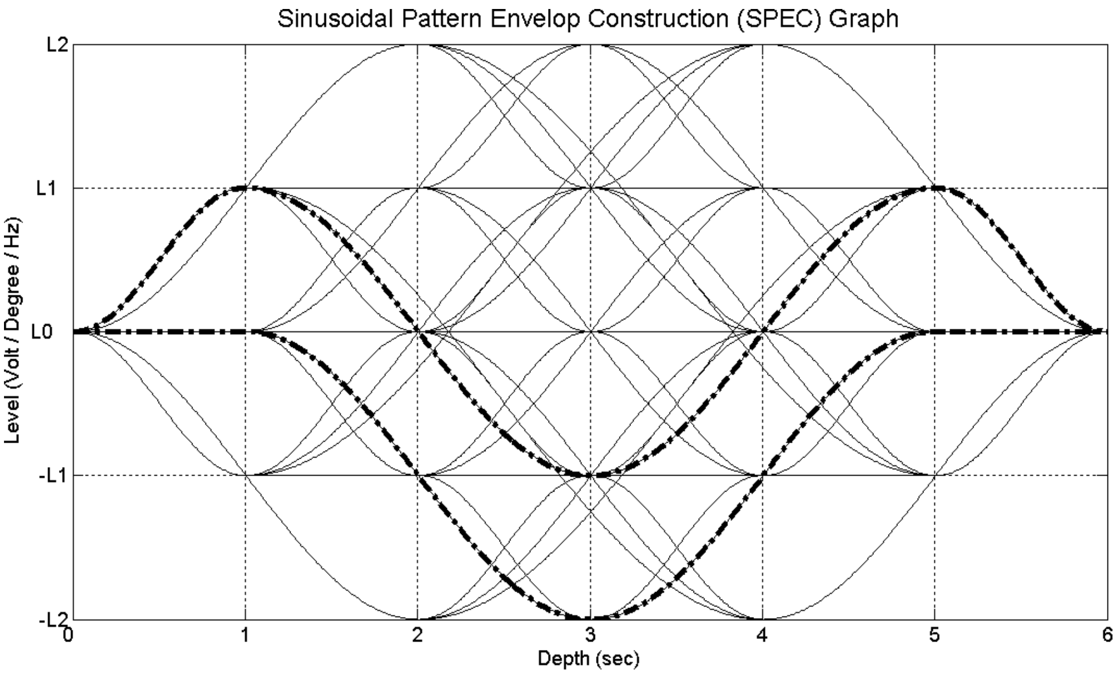

Figure 3.7 : One alternative route for P in SPEC space.

$$A_{i,j}(t) = d \times \sin(t) \begin{cases} d = \sum_{s=j}^{j+d} [A_s]_{1\times D} \\ d > 0 \Rightarrow t = \left[-\pi/2, \pi/2\right] \\ d < 0 \Rightarrow t = \left[\pi/2, 3\pi/2\right] \\ \Delta t = \dfrac{\pi}{Q} \times \dfrac{D}{|d|} \end{cases} \tag{3.4}$$

where, D, d and Q represent the number of depth in SPEC space, the total value of the same signs in an order for $[A_s]$ and the number of samples per signal pattern (waveform), sequentially. Δt shows the step of the change in the time space. The reason of π in the definition of Δt is coming from the half sinus function. Matrix A_s presents three different conditions, which varies between -1 (half sinus form in between +π/2 and +3π/2) and +1 (half sinus form -π/2 and +π/2) with respect to one of the envelope combinations shown in Figure 3.6. If there is no change between two depth levels, the value of A_s at the respecting matrix member defined as 0. For instance, matrix $[A_s]_{1x6}$ for route 1 in Figure 3.6 equals to [1 1 -1 -1 -1 1]. For this example, since first two signs are the same and then it changes for 3 signs, d values equal to +2, -3 and +1 in sequence. In this example, $A_{i,j}(t)$ has 3 different cases. The first case (d=+2) uses the sinus form changes between –π/2 and +π/2 as mentioned in equation 1. This points the first two depth of route 1 in Figure 3.6. On the other hand, the second case (d=-3) uses the signal from between +π/2 and +3π/2. This is also mentioned in Depth between 3rd and 5th layers at the same figure. This approach is also valid for F and P features. Therefore, $f_{i,j}(t)$ and $\Phi_{i,j}(t)$ can be defined as $A_{i,j}(t)$ in equation 1. After these variables are defined, the communication signal form for j^{th} signal pattern (waveform) from the i^{th} glossary is defined as

$$\begin{aligned} &S_{i,j}\left(A_{i,j}(t), f_{i,j}(t), \Phi_{i,j}(t), t\right) = \\ &\quad \left[\left(A_0 + \left(A_{i,j}(t) \times \Delta A\right)\right)\right] \times \sin\left[2\pi\left(f_0 + \left(f_{i,j}(t) \times \Delta f\right)\right) \times t + \left(\Phi_{i,j}(t) \times \Delta\Phi\right)\right] \\ &\qquad\qquad\qquad \text{for } i = 1 \ldots M;\ j = 1 \ldots N_{(i)} \end{aligned} \tag{3.5}$$

where f_0 is the central frequency, A_0 is the mean value of amplitude interval, ΔA, Δf and $\Delta\Phi$ are allowed limitation on amplitude, frequency and phase change for SPEC y-axis (Level). In the initial condition of t=0 (at L0), which corresponds to (0, 0, 0, 0) coordinate on the SPEC space. The formula for this basic signal pattern is simplified into the following equation and it represents the carrier signal of the constructed pattern.

$$S_0(t) = A_0 \times \sin(2\pi \times f_0 \times t) \tag{3.6}$$

The constructed signal patterns $S_{i,j}$ (waveforms) could be used to express the pattern glossary space mathematically, this matrix is named as Γ, and the equation is given as below,

$$\Gamma = \begin{bmatrix} \Gamma_1 \\ \Gamma_2 \\ : \\ : \\ \Gamma_M \end{bmatrix} = \begin{bmatrix} S_{1,1}(t) & S_{1,2}(t) & \ldots & \ldots & S_{1,N_{(1)}}(t) \\ S_{2,1}(t) & S_{2,2}(t) & \ldots & \ldots & S_{2,N_{(2)}}(t) \\ : & : & : & : & : \\ : & : & : & : & : \\ S_{M,1}(t) & S_{M,2}(t) & \ldots & \ldots & S_{M,N_{(m)}}(t) \end{bmatrix} \tag{3.7}$$

where each row represents a group of $N_{(i)}$ signal pattern inside the respecting glossary. Due to the adaptive modulation approach, the glossary set Γ_i is chosen from the Γ matrix for the different channel conditions and bit level. However, the length of the data frame is always protected during the communication. This is required for synchronization between the transmitters and the receivers besides satisfying the continuation of the communication signal. Each communication pattern is the consequence of the signal patterns from one of the M glossaries. The number of signal pattern $N_{(i)}$ is determined by encoding of the binary data to be transferred. For instance, if Γ_i is 4-bit glossary, $N_{(i)}$ is set to 16.

Due to the above conditions and approaches, SPEC does not need to use any additional clustering and/or classification algorithm anymore. It lists all possible routes. According to the correlation matrix for these routes, it selects the most uncorrelated choices with the needed quantity (i.e. it selects 16 most uncorrelated routes for 4-bit glossary construction).

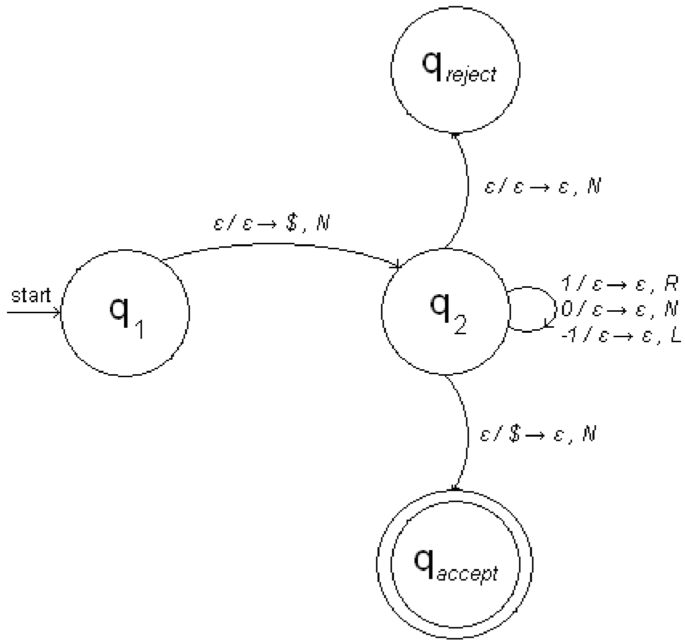

Figure 3.8 : State diagram for the valid combinations of SPEC algorithm.

Figure 3.8 represents the state diagram designed by Turing Machine rules for SPEC algorithm. This machine is used to determine the valid combinations of the sinusoidal pattern envelopes. First, the symbol $ inserts the tape at null input. The tape head moves right for “+1” and left for “-1”. If the input is “0”, the tape head does not move. Whole symbol is fed into the machine. At the end, if the tape head points initial position of the tape, namely $, machine pulls the symbol and goes to “accept” state. In this case, the machine accepts the sequence and adds it to the valid list. Otherwise, the machine ends with “reject” state and the sequence is not added to the valid list. According to this state diagram, “+1” points the increment in the level with the quarter sinus change, oppositely, “-1” represents the decrement in the level with the quarter sinus change and the means for “0” is the staying on the existing level. Summation of the arrays in any route from left to right should be zero. The route begins with zero, ends with the initial node so the state diagram accepts the sequence. The state machine ends with "accept" state if the sequence satisfies the Zero DC component and continuity conditions mentioned above. This machine proves the applicability of PBCS in any hardware and software platform.

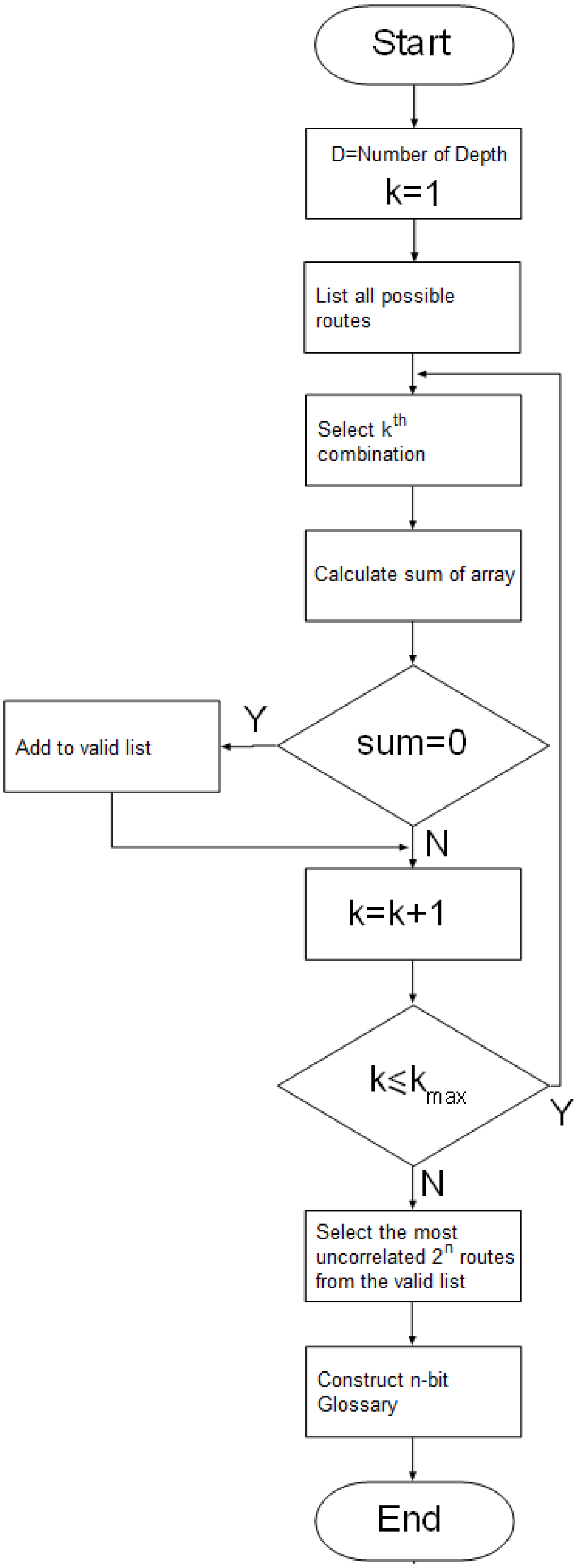

Figure 3.9 : Flow chart for SPEC algorithm.

All possible routes are constructed by three actions (minus quarter sinus, plus quarter sinus and constant). The mathematical expression for the list of all possible routes shown in Figure 3.5 is given in following formula,

$$\left[e^{*}\right]_{3^{\mathrm{Dh}}\times \mathrm{Dh}} \quad ; e=\{-1, \quad 0, \quad 1\} \tag{3.8}$$

where, “Dh” represents the depth of the SPEC space. The matrix above shows all possible alternatives for three signal features (amplitude, frequency and phase). The first dimension of the array shows the number of all alternative routes and the second dimension of it represents the depth of SPEC space. The machine shown in Figure 3.8 eliminates the non-proper alternatives and lists the valid sequences in this space.

Figure 3.9 shows the glossary construction flow chart for the proposed envelope construction. It explains whole process systematically.

The test signal constructed by the glossary patterns should satisfy the communication masks for narrowband and spread spectrum communication. These masks are defined by some standards. The tests for the future works are performed for SPEC algorithms with IEEE 802.11g mask for this standard. The simulation results for future works after the usage of this filtering procedure are located in section 4.

3.2.2 Pattern based encoding

Glossary construction is performed with offline process. Each pattern on the glossary presents one binary word with different length. The encoder reads the binary word from the information source and it matches this word with the related pattern from glossary space. The patterns are concatenated in the buffer (sequencer). The concatenated signal packet is transmitted through antenna front-end.

3.3 Communication Channel Model

A communication channel refers either to a physical transmission medium (i.e. wired communication system) or to a multiplexed connection over a shared medium, such as a radio channel, in communication systems.

Physically, a channel can be modeled by trying to calculate the physical processes. They modify the transmitted signal. For instance, the channel model can be defined by calculating the reflection off every object in the environment in wireless communications. A sequence of random numbers might also be added in to the transmitted signal to simulate the received signal form in the receiver.

Statistically, communication channels are usually modeled with three parameters consisting of an input alphabet, an output alphabet and for each pair of input and output elements a transition probability. This probability parameter refers to BER in this study to show the performance of the systems.

Statistical and physical modeling can be combined. For example, the channel is often modeled by a random attenuation of the transmitted signal, followed by additive noise in wireless communication systems. The attenuation term is a simplification of the underlying physical processes and captures the change in signal power. The noise in the model captures external interference and/or noise in the receiver. If the attenuation term is complex, it also describes the relative time a signal takes to get through the channel. Previous measurements or physical simulations decide the statistics of the random attenuation.

A channel model may either be digital or analog. In a digital channel model, the transmitted message is modeled as a digital signal at a certain protocol layer. A simplified model replaces underlying protocol layers, such as the physical layer transmission technique. The model may reflect channel performance measures such as bit rate, bit errors, latency/delay, and delay jitter, etc. Some of digital channel models can be listed as:

- Binary symmetric channel (BSC),
- Binary bursty bit error channel model,
- Binary erasure channel (BEC),
- Packet erasure channel,

The transmitted message is modeled as an analog signal in an analog channel model. The model can be a linear or non-linear, time-continuous or time-discrete, memoryless or dynamic, time-invariant or time-variant, baseband, pass-band (RF signal model), real-valued or complex-valued signal model. The model may reflect the following channel impairments:

- Noise model,

    - Additive White Gaussian Noise (AWGN) channel, a linear continuous memoryless model
    - Phase noise model
- Interference model, for example cross-talk (co-channel interference) and inter-symbol interference (ISI)
- Distortion model, for example a non-linear channel model causing inter-modulation distortion (IMD)
- Frequency response model, including attenuation and phase-shift
- Group delay model
- Modeling of underlying physical layer transmission techniques, for example a complex-valued equivalent baseband model of modulation and frequency response
- Radio frequency propagation model, for example
    - Log-distance path loss model
    - Fading model, for example Rayleigh fading, Ricean fading, log-normal shadow fading and frequency selective (dispersive) fading
    - Doppler shift model, which combined with fading results in a time-variant system
    - Ray tracing models, which attempt to model the signal propagation and distortions for specified transmitter-receiver geometries, terrain types, and antennas
    - Mobility models, which also causes a time-variant system

The communication medium in PBCS uses analogue signal patterns to transmit the binary sequence through communication channel. In this study, PBCS is performed by the utilization of noise model in the simulation experiments. AWGN is a channel model in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density (expressed as watts per hertz of bandwidth) and a Gaussian distribution of amplitude.

The attenuation term does not affect the performance of PBC as much as noise model, because the recognition performance of ANN directly depends on the phase and frequency changes. The pre-trained ANN can recognize the signal patterns, if only the amplitude of signal attenuates. In this case, the attenuation in the amplitude can be ignored. Another most common channel model is fading effect. The multipath model definition and the solution advices have already are given in section 2 for the future works.

3.4 Receiver Structure

As presented in the right hand-side of Figure 3.2, the distorted signal in the communication medium arrives to the antenna block including front-end. It goes through a synchronization process. The analog signal is sampled by sample and hold block as shown in Figure 3.10. The output of this block is fed into the ANN as an input. Since ANN is pre-trained for the patterns, it can recover the distorted signal into known patterns, and outputs the associated bit sequences. The “Bit Sequencer” layer concatenates these bit sequences in order to come up with the original information.

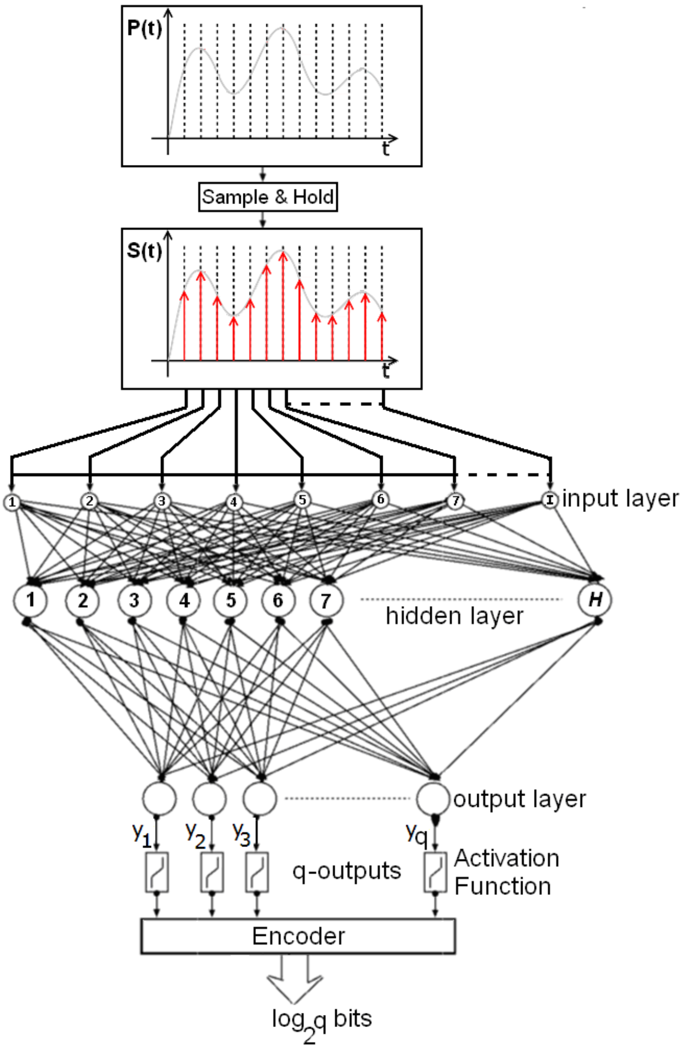

Figure 3.10 : Decoder / Demodulator structure in PBCS.

3.4.1 ANN construction

The recognition layer has the most crucial effect on the performance of the system. This layer includes the training and the test processes. In the training phase, the ANN is trained offline according to the glossary patterns. Glossary pattern set is used for training. The signal pattern in this set is distorted in communication medium. The set of these distorted signal patterns is called a test set. Three different architectures are proposed for ANN structure of PBCS. These are compared with the simulation tests and the most proper structure is determined according to test results.

3.4.1.1 Multilayer perceptron

Multilayer Perceptron (MLP) is the first defined platform in artificial neural network concept in the literature. Some cognitive scientists and neuroscientists construct models that simulate the natural neural networks in the brain (Posner, 1989). Then, different researchers develop this concept. The semiconductor technology has supported the parallel processing in 1980's and this process has affected the popularity of ANN. It has become very famous research area late 1980's. In this study, the form of MLP defined by Haykin (1999) is taken as a reference. Since ANN achieved successful results in complex classification and pattern recognition problems, it is selected to recognize the signal patterns (waveforms). These signal patterns are constructed based on the RF signal construction principles as explained before.

MLP structure includes some hidden layers. These layers have nodes named neuron. Neural networks are implemented solving simple nonlinear or linear processing functions. Here, neurons interconnect with other ones forming complex processing networks. These networks are trained by adjusting the interconnection branch loads (synapse weights) through training algorithms (Zurada, 1992, Kohonen, 1980). Conventional expert systems interpret the knowledge as a set of rules. In contrast to them, artificial neural networks learn from given examples and establish their own set of rules (Rojas, 1996, Schalkoff, 1997).

A decisive property of ANN is its learning ability. Learning of ANN is realized through changing the connection weights of network. This process depends on input and related outputs. There are two types of learning strategies in literature. They can be classified as supervised and unsupervised learning (Efe and Kaynak, 2000, Alpaydin, 2004). The basic difference between them is the existence or nonexistence of desired outputs during learning procedure. The expectation from ANN is to verify information and to give as low as possible error at the output. This study uses the learning capability of ANN with supervised learning approach. It decodes the distorted patterns at the glossary. This means that the training set includes the glossary patterns and the test set contents distorted patterns.

ANN refers to “black-box” in some studies; it aims to determine any function between input and output values. The difference between the ANN output and the actual output is fed into the system using the back-propagation algorithm. After a number repetition, a function can be formulated between the input and the output. The number of repetition is determined with respect to the user defined finishing criterion. This process requires the computing of the error derivative of the weights. In other words, it must calculate how the error changes as each weight is increased or decreased slightly. The back propagation algorithm is the most widely used method for determining the error derivative of the weights.

Many methods are developed for back-propagation algorithms in the literature. Since the most applicable to the proposed system, “Steepest Descent” is chosen in this study. This method based on minimization of the quadratic cost function by tuning the network parameters (Rumelhart et al., 1986). The mathematical expression is this algorithm as below.

$$w_{ji}^{(k+1)} = w_{ji}^{(k)} - \eta \frac{\partial \varepsilon^{(k)}}{\partial w_{ji}^{(k)}} \quad \textit{for } j=1,\dots,q\ ;\ \textit{for } i=1,\dots,H \qquad \textbf{(3.9)}$$

where, w represents the weights of the ANN, k is the number of iterations, ε is error between the output of ANN and the actual value, η is the learning rate.

$$\eta^{i} = \begin{cases} \eta^{i-1} \times 0.995 & \text{if } \dfrac{\partial \varepsilon}{\partial w} < 0 \\ \eta^{i-1} + \left(0.0001 \times \varepsilon\right) & \text{if } \dfrac{\partial \varepsilon}{\partial w} > 0 \end{cases} \qquad \textbf{(3.10)}$$

Learning rate can be selected as a constant or an adaptive coefficient. It is updated by the adaptive approach (additive increment and multiplicative decrement) in the decision layer of PBCS design. The adaptive method is called as “additive increment and multiplicative decrement". The formulization of this procedure is in equation 3.10.

Another important aspect in ANN design is to determine the number of hidden layers and of neurons located in these layers. From the low complexity and high applicability perspectives, a single hidden layer is chosen in PBCS design. The number of neurons in this layer is determined by the cross-validation method. On the other hand, the number of neurons at input layer is set as equal to the number of samples for each pattern. The number of neurons at the output layer is selected as equal to the number of patterns in the glossary.

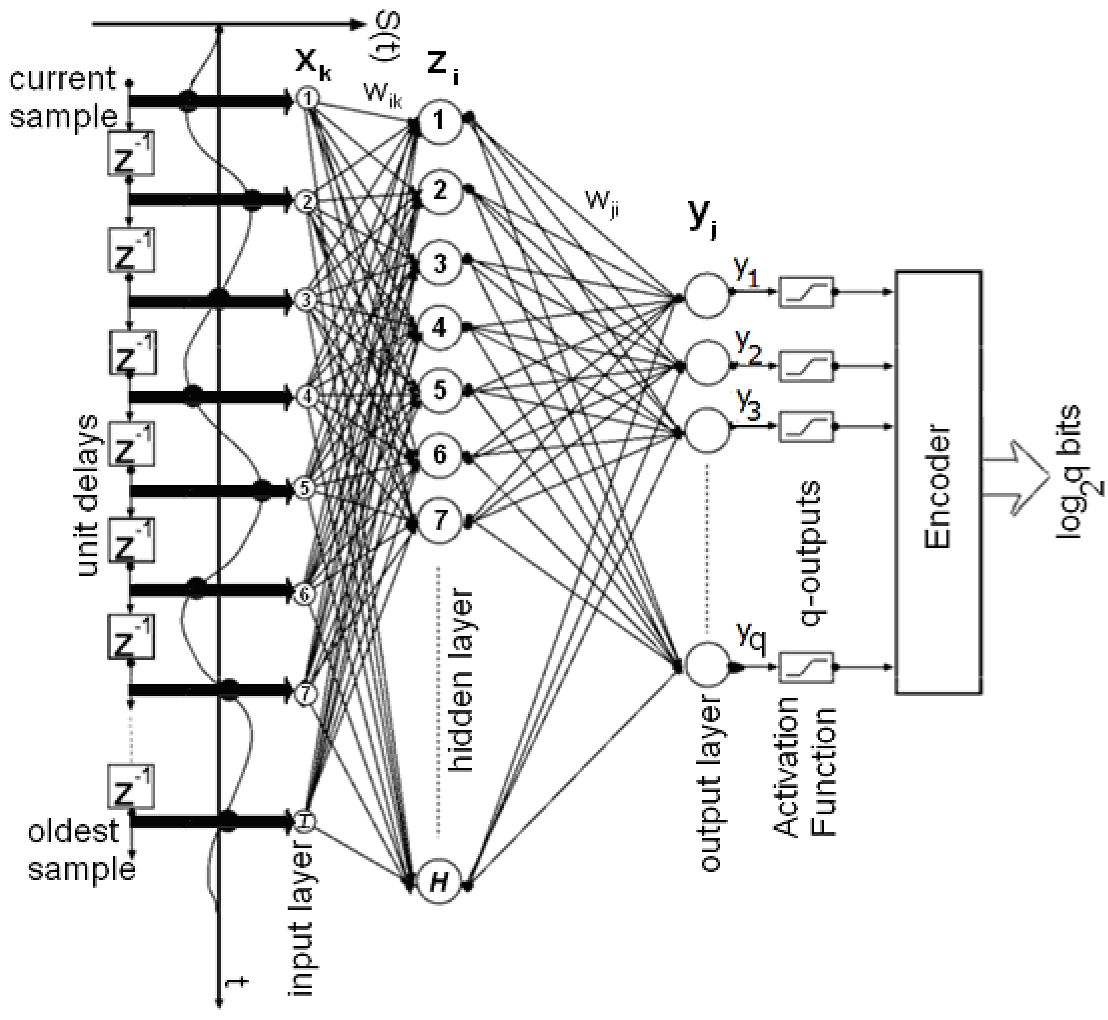

Figure 3.11 : The general view of MLP architecture.

The constructed signal pattern S(t), is sampled by the unit time delay. Each sample corresponds to the neurons of the input layer. The number of neuron in the hidden layer is determined by the cross-validation in offline process. The outputs $y_1, y_2, \ldots, y_q$ indicates recognition levels between the glossary and the applied signal in the q-tap frame. The encoder block in this figure encodes the output of ANN. The output of ANN is converted into respecting binary signal sequences in the order of glossary. The binary sequences at each frame instant are added to recover all the information at the receiver where these frames were encoded by ordering the patterns at the origin. ANN structures with these values are represented in Figure 3.11.

The error in ANN is calculated in terms of mean squared error (MSE). The aim is to decrease this error at each training process and finally to bring it below the user-defined threshold. The resulting error value is given in the equations below,

$$e_j(n) = d_j(n) - y_j(n) \quad for\ j=1,\ldots,q \tag{3.11}$$

where, *d* is the actual response and *y* is calculated response. The difference between them signed as error (*e*). *q* represents the number of neuron at output layer. MSE can be calculated as

$$\varepsilon(n) = \frac{1}{\tau}\sum_{n=1}^{\tau}\sum_{j=1}^{q} e_j^2(n) \tag{3.12}$$

where, *e*, τ and ε denote error of neural network at output, total number of patterns in the training set and mean squared error sequentially.

The synaptic weights are adjusted to minimize the quadratic cost function. However, this process is different between the layers. For example, the adjustment of synaptic weights between hidden layers and output layer is given by the equation 3.13 and 3.14,

$$\delta_j(n) = \Psi'\left(\sum_{j=1}^{q} w_{ji}(n) z_i(n)\right) e_j(n) \qquad \begin{cases} \text{for j} = 1,\ldots,\text{q} \\ \text{for i} = 1,\ldots,\text{H} \end{cases} \tag{3.13}$$

$$\Delta w_{ji}(n) = \alpha\, \Delta w_{ji}(n-1) + \eta \delta_j(n) z_i(n) \qquad \begin{cases} \text{for j} = 1,\ldots,\text{q} \\ \text{for i} = 1,\ldots,\text{H} \end{cases} \tag{3.14}$$

where, ψ is activation function, which will be explained below in detail. δ shows the synaptic weights. *H* represents the number of neuron at hidden layer. Furthermore, the adjustment of synaptic weight coefficients between the input layer and the hidden layer is given by equation 3.15,

$$\delta_i(n) = \Psi'\left(\sum_{i=1}^{H} w_{ik}(n)u_k(n)\right)\delta_j(n)w_{ji}(n) \qquad \begin{cases} \text{for } j = 1,\ldots,q \\ \text{for } i = 1,\ldots,H \\ \text{for } k = 1,\ldots,I \end{cases} \tag{3.15}$$

where, I represents the number of neuron at input layer and η indicates learning-rate. Designer can design it as constant or adaptive. In case the network does not converge, Rumelhart et al. (1986) has generalized the formula. It uses to determine weight coefficients including α momentum parameter. This formula is given in equation 3.16.

$$\Delta w_{ik}(n) = \alpha\, \Delta w_{ik}(n-1) + \eta\delta_i(n)y_k(n) \tag{3.16}$$

The activation function is effective on the sensitivity of the output layer. There are many activation functions in the literature. Three main activation functions are called sigmoid, tangent hyperbolic and linear. They are drawn in Figure 3.12.

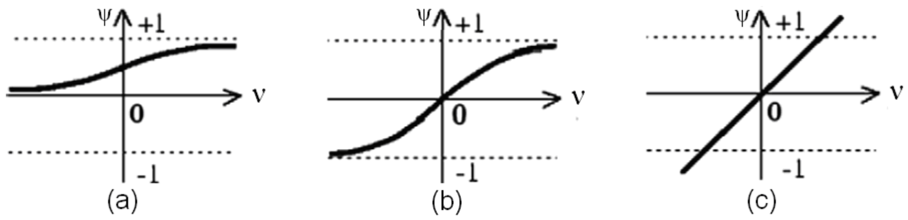

Figure 3.12 : (a)Sigmoid, (b)Tangent Hyperbolic, (c)Linear.

Due to the appropriation of upper and lower limit and its transition function, the sigmoid function is selected as the activation function. The problem in this system is to determine which pattern has entered by the input layer. The output corresponding to the input pattern should be one, and the rest should be zero. The most proper activation function for this purpose is the sigmoid as given in equation 3.17.

$$\Psi(\nu) = 1/(1+\exp(-\lambda \times \nu)) \tag{3.17}$$

This function carries the problem to determine a constant ("λ") at the formula. This parameter affects the performance of the ANN directly. It can be determined by the test experiments.

3.4.1.2 Radial based neural network

Radial Based Neural Network (RBNN) is any type of artificial neural network model. It uses radial basis functions (RBF) at hidden layer (Moody and Darken, 1989). Radial basis functions are useful techniques for interpolation in multidimensional space problems. RBF has built into a distance criterion with respect to a centre.

RBNN has two layers of processing. The input is mapped onto each RBF (generally Gaussian function) in the hidden layer. In regression problems, the output layer is a linear combination of hidden layer values representing mean predicted output. The interpretation of this output layer value is the same as a regression model in statistics. In classification problems, the output layer is typically a sigmoid function of a linear combination of hidden layer values, representing a posterior probability.

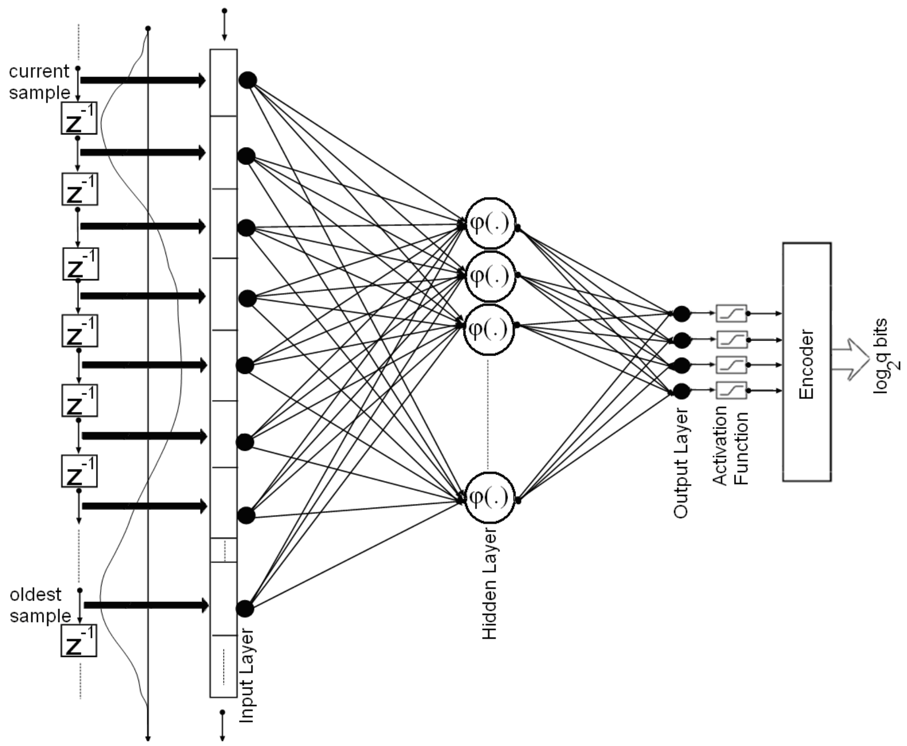

Figure 3.13 : General view of RBNN and WBNN architectures.

RBNN has the advantage of not suffering from local minima in the same way as MLP. The adjusted parameters in the learning process are the linear mapping from hidden layer to output layer. Linearity ensures that the error surface is quadratic and therefore has a single easily found minimum. In regression problems, this can find the minimum in one matrix operation. On the other hand, RBNN has the disadvantage of requiring good coverage of the input space by radial basis functions.

RBF centers are determined with reference to the distribution of the input data, but without reference to the prediction task. As a result, representational resources of the input space are irrelevant to the learning task. RBNN architecture is very similar to MLP, except hidden layer function. RBF of a hidden layer is determined as the following formula

$$\varphi(x) = e^{-\frac{(x-\mu)^2}{2\sigma^2}} \tag{3.18}$$

where, μ and σ are the mean and standard deviation values of the input set sequentially. RBNN model is located in the Figure 3.13.

3.4.1.3 Wavelet based neural network

Zhang and Benveniste (1992) has proposed an alternative structure to the feed forward neural networks for approximating arbitrary nonlinear functions. This alternative structure is called Wavelet Based Neural Network (WBNN). It has similar structure with RBNN. However, WBNN uses different function at hidden layer. The basic idea behind WBNN is to replace RBF function of RBNN at hidden layer with any wavelet transform function. It is selected as Mexican Hat and given in the following formula,

$$\varphi(x) = \left(\frac{(x-\mu)^2}{\sigma^2} - 1 \right) e^{-\frac{(x-\mu)^2}{2\sigma^2}} \tag{3.19}$$

3.4.2 Synchronization of signal

The received signal from RF front-end should be ordered in the receiver side. While RF signal is transmitting trough communication medium, it reflects from different surfaces and materials. When it comes to RF receiver with this form, the receiver needs to synchronize the distorted and combined signal form to decode the information. After this process, the synchronized signal is given to ANN as an input. The success rate of this part directly affects overall system performance.

The disorder in the signal caused by the reflection and delay in communication channel is a reason of multipath effect. This effect can be estimated by several methods by the utilization of traditional methods. The important problem in this situation is to determine the first and last points of any signal pattern. The solution advices for this problem will be given in this section.

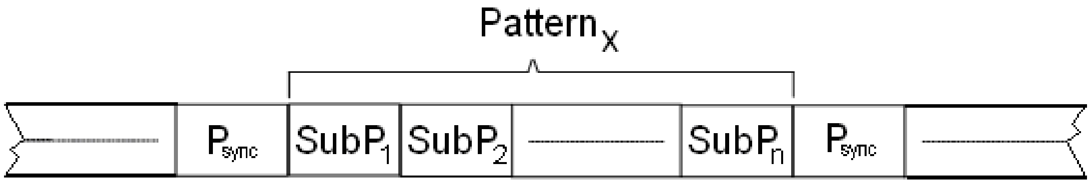

Figure 3.14 : Synchronization pattern approach for synchronization block.

The synchronization block is located in the receiver before intelligent layer. Its mission is to determine the beginning point of each pattern. One of the alternatives for solution is to use preamble pattern. This pattern is assigned as synchronization pattern (P_{sync}). It is located in front of each pattern of transmitted signal. ANN first recognizes this pattern (P_{sync}) and it determines the beginning point of glossary pattern. The proposed method is drawn in Figure 3.14. The “$SubP_x$” in this figure represents the sub-patterns. This concept corresponds to depth in SPEC approach.

The alternative method is to sense the change in the signal. ANN has this capability. Therefore, additional ANN can be put before the intelligent layer. The sampled analogue signal crosses through ANN input layer with time delay. If the designed ANN has one output, it decides a valid pattern. Unless the window correspond any valid pattern, it ignores the input and tests one unit time shifted pattern. Inspired by the study about Phase-Locked Loop Neural Network (PLLNN) (Hoppensteadt and Izhikevich, 2000), this solution advice is possible to implement in hardware platform. This work combines the PLL and ANN features to sense and determine the phase changes. The input layer of ANN structure is designed in a bit different manner from MLP. Although, this work proves that PLLNN is one of the important solutions for phase change determination and synchronization, according to comments of its authors, it needs to some improvements and optimizations.

3.4.3 Error correction

Another mandatory feature for the communication system is an error correction. In this study, recursively PBCS usage is proposed for error correction. Each PBCS block determines character, word, sentences and contents respectively. This feature provides overall system with the error correction. The proposed process is shown in Figure 3.15.

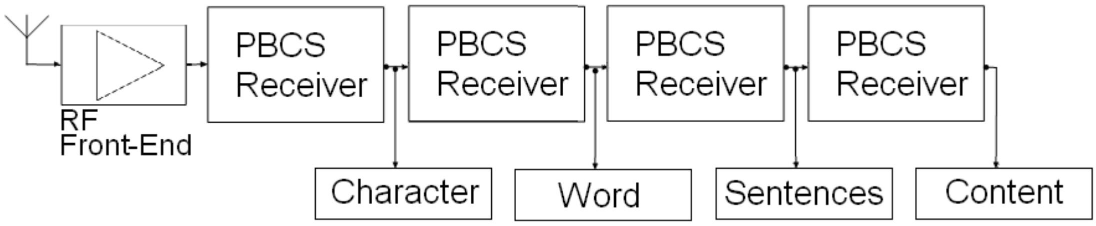

Figure 3.15 : Error correction structure.

3.4.4 Pattern based decoding

Antenna front-end of receiver is adjusted related communication channel. The distorted signal is captured from the communication medium via tuned antenna. “Synchronizer” eliminates multipath effect in fading channel and determines the pattern frame. The distorted pattern with only AWGN is fed into ANN as an input. The offline trained ANN decodes the pattern. It assigns the pattern to the related binary sequence. “Bit sequencer” concatenates these binary words in an order. It gives the output as the reconstructed original information.

The transmitter and receiver structures of PBCS were described in detail. These are formed in line with unique algorithms in order to enable a more dynamic utilization for the communication medium. PBCS is designed to use in time-invariant and AWGN channel model for CR applications in this thesis. However, the boundaries of this work can be extended with some innovative approaches. For instance, the algorithms of pattern envelope construction are the main innovative parts of this thesis. These approaches depend on linear and sinusoidal changes in the signal features. Since these varying methods are the fundamental approaches, they were selected for this study. Nevertheless, more complex varying methods can be defined as a future work such as Gaussian, Raised Cosine or Hermite pulse etc. The space defined in SPEC algorithm is the most important feature of PBCS in terms of flexibility and it can be used for the different waveform envelope functions. The expected result is to obtain the more flexible and manageable platform for communication lines.

The performance evaluation of the proposed system is located in the next section. First, the recognition capability of ANN is shown. In the second phase, different construction algorithms of signal pattern are developed to find out the most proper algorithm in terms of spectral bandwidth consumption and the robustness to noise suppression. The ANN design is performed with the selected construction algorithm.

A performance measure function that refers to minimization of the area under the BER vs. E_b/N_0 graph is described and PBC is compared with its equivalent benchmark works. For this purpose, the semi-hardware platform is prepared and the implementation of PBC for baseband application is performed. The result for this study is also located in the end of the following section. The multipath effect and the solution advice are defined for the future works. The following section will cover and analyze the simulation / semi-hardware platform results regarding the design process of PBCS.

4. PERFORMANCE EVALUATION OF PBCS

In this thesis, a new modulation and coding approach is proposed for the adaptive communication based system. This proposed system depends on the pattern recognition and it uses the neural networks in the physical layer. The main purpose is to utilize the pattern recognition capability of the neural networks to improve the adaptive communication performance. This approach provides the more robust system against to the noise effect and the higher data transfer rate on the communication channel. Many studies are performed to evaluate the performance of the proposed system. These tests are simulated in MATLAB simulation tool. First, the predefined pattern (waveform) sets, namely glossary, are used for showing the recovery performance of ANN. In the second phase, the different glossary construction methods are proposed and developed to discuss the effect of different glossary utilization. These developed methods are evaluated step by step in terms of pros and cons. The comparison between them is presented by the simulation results. According to them, the most proper algorithm of signal pattern construction is selected. ANN is designed for the constructed glossaries. The comparison between the most related methods and PBC is given in this section. PBC behavior in baseband communication is realized at semi-hardware platform to prove the applicability of PBC in real-time applications. The flexible structure of PBC and manageable SNR capability is supported by the simulation and semi-hardware platform results. Finally, the performance analysis of multipath effect is simulated with the proposed approach for constructing the basis of future works.

4.1 Pattern Recognition Capability of ANN

The pattern recognition capability of neural networks is well-known feature of it. However, the performance of MLP should be tested for the communication signal patterns. This test shows the applicability of PBC. For that reason, many tests are performed to evaluate the recoverability of MLP. The simulation tests are implemented either single distorted patterns or two combined signal patterns. The first part of the tests gives an idea about the recognition capability of MLP for communication signal patterns. The rest of them show the applicability of PBC for multiple accessing on communication channel. These tests and the results are published as a conference paper (Ustundag and Orcay, 2008).

MLP is designed with two hidden layer at the first part of the study. Due to the signal patterns are sampled with 300 samples, the number of neuron at the first hidden layer is determined as 300. The number of neuron at second hidden layer is selected as the half of the first one. The output layer represents the number of output, so the number of neuron is set to the number of output. Less amount of neurons are preferred in the further steps of the investigation due to implementation factors. The learning rate is chosen constant as $w^{(1)}=0.6$, $w^{(2)}=0.4$ and $w^{(3)}=0.2$ within the order of layers. The momentum coefficient, α, is set to 0.5 in the simplified example. The constant for the sigmoid, λ, is set to 0.6. The described neural network is presented in Figure 4.1.

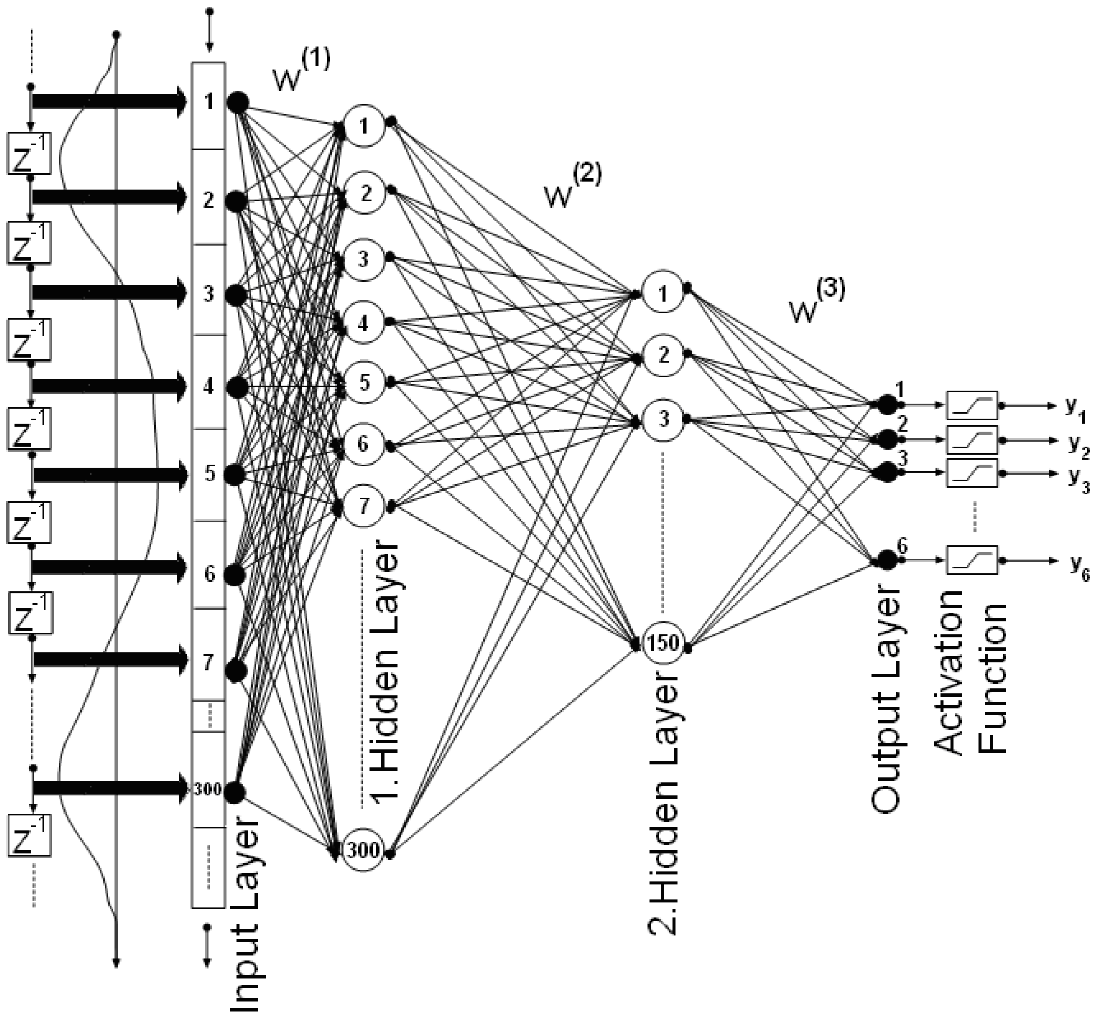

Figure 4.1 : The utilized neural network structure (two hidden layer).

Each test pattern is constructed with three sub-patterns in this part of study. The signal patterns are generated with the bandwidth limited switching between three different frequency and amplitude levels. These sample signal patterns are distributed over a larger spectrum than the strictly band limited real case. The primary form of base signals is shown in Figure 4.2. Two features of communication signal (frequency and amplitude) are chosen in three different levels. These levels are listed in Table 4.1.

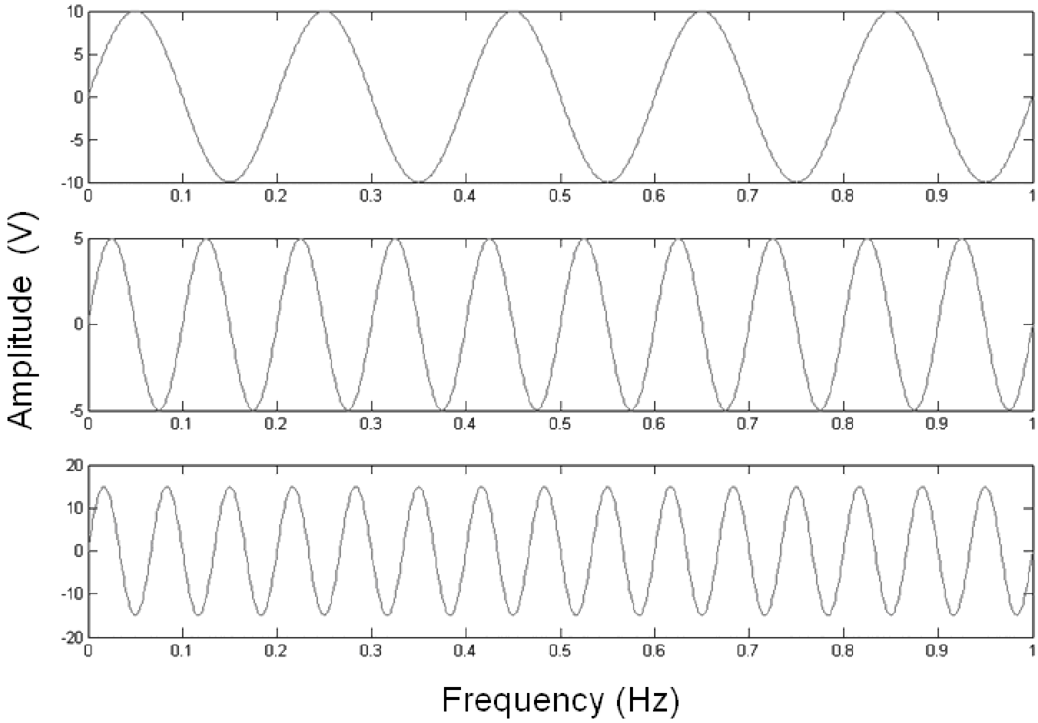

Figure 4.2 : Base signals.

Table 4.1: Base signals specifications.

	A1	A2	A3
Frequency (Hz)	5	10	15
Amplitude (V)	10	5	15

These base signals are used to construct sample signal patterns. All combinations are constructed with them. These are collected in one set. This set has six sample patterns and is shown in Figure 4.3. The patterns in this figure are thought to MLP. Test set is derived from this training set. The Additive White Gaussian Noise (AWGN) is applied to the training patterns. The distorted patterns are used for testing. This test set is shown in Figure 4.4. The noise rate is increased until the recognition based decoding becomes unsatisfactory. When the SNR value decreased around -9.5 dB, MLP begins to be unsuccessful to recover data. This level for the pattern recognition in communication signals proves that MLP can be used for communication signal recognition.

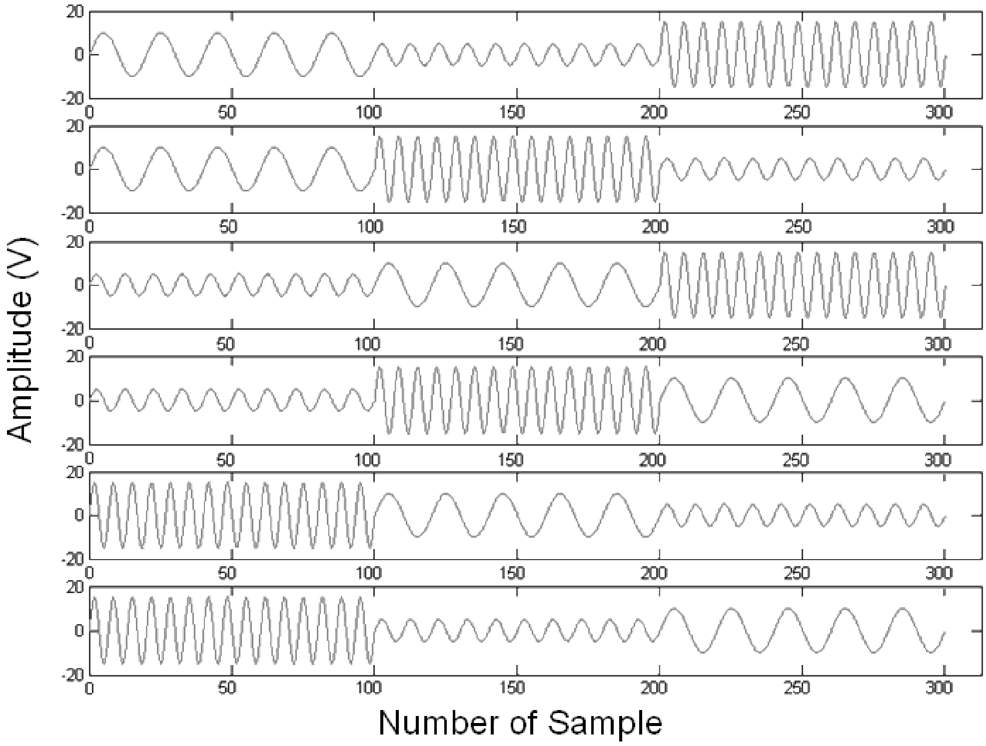

Figure 4.3 : The signal patterns in glossary (training set).

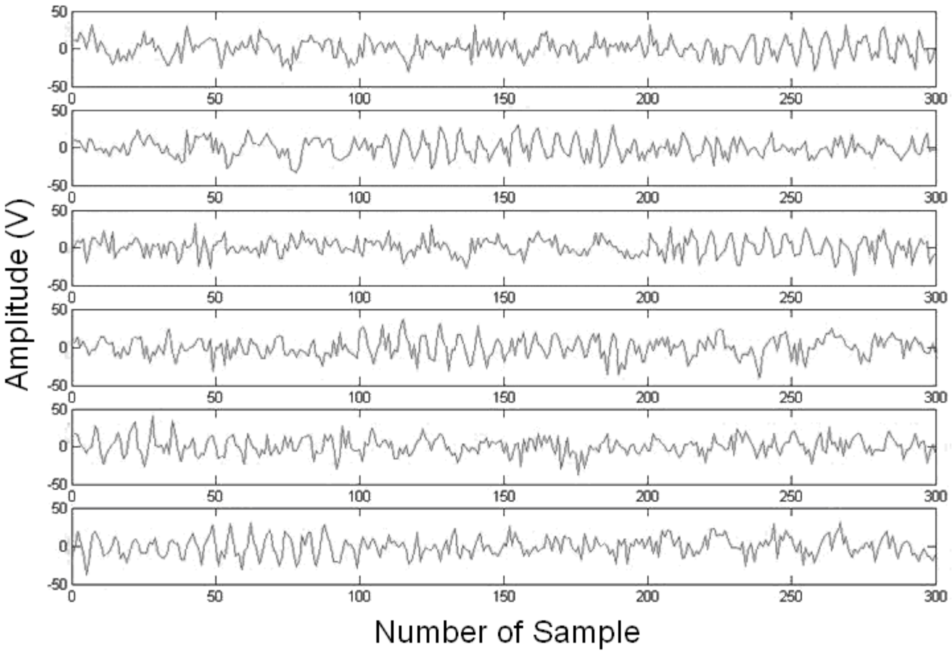

Figure 4.4 : The distorted signal patterns (test set) (SNR = -9.5 dB).

The recognition performance at -9.5 dB SNR value of MLP is located in Table 4.2. This table shows the maximum values diagonally. It means that MLP recognizes the test patterns successfully even in high noise suppression.

Table 4.2: The normalized success rate of MLP for the distorted signal patterns.

		Original Signals					
		S1	S2	S3	S4	S5	S6
Predicted Signals	Y1	1.000	0.128	0.082	0.000	0.165	0.244
	Y2	0.315	1.000	0.087	0.305	0.000	0.293
	Y3	0.000	0.303	1.000	0.282	0.098	0.142
	Y4	0.016	0.000	0.321	1.000	0.128	0.137
	Y5	0.000	0.470	0.268	0.635	1.000	0.707
	Y6	0.372	0.032	0.205	0.284	0.000	1.000

The other tests are performed for two combined signal patterns. Some of the combinations for the signal patterns are located in Figure 4.5. This pattern set is used for test set. The recognition rate of MLP is located in Table 4.3. The maximum values are obtained for the main contents of the test signal patterns. For example, Y1 contents the signal patterns S1 and S2. According to Table 4.3, the row for Y1 shows the maximum values at S1 and S2 columns; this means that PBC can be used for multiple-access purpose in communication systems.

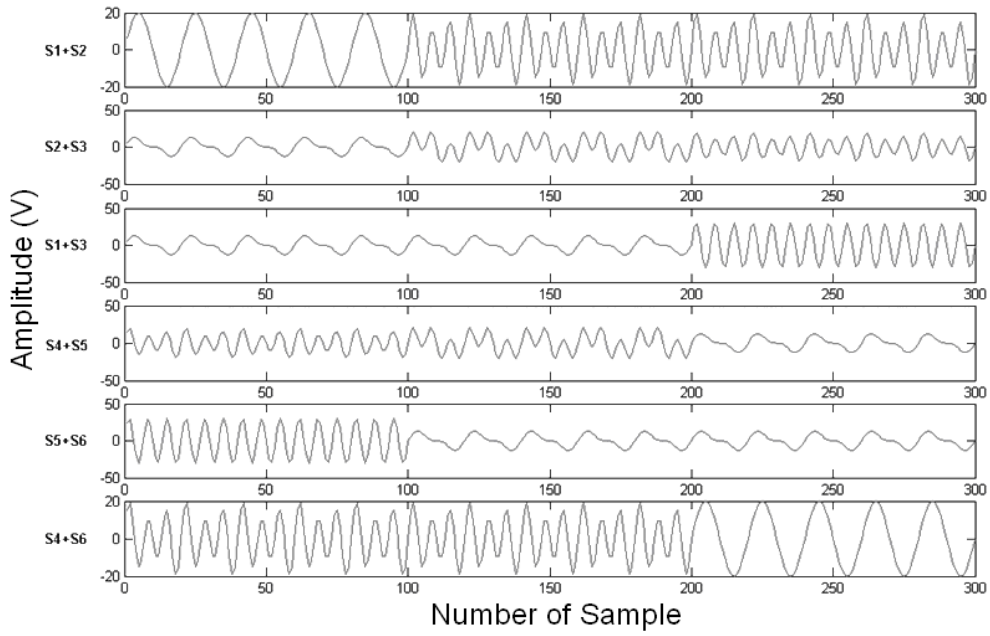

Figure 4.5 : The combined signal patterns (test set).

Table 4.3: The normalized success rate of MLP for the combined signal patterns.

		Original Signals					
		S1	S2	S3	S4	S5	S6
Predicted Signals	Y1 (S1+S2)	0.382	1.000	0.029	0.106	0.000	0.056
	Y2 (S2+S3)	0.000	1.000	0.542	0.354	0.091	0.052
	Y3 (S1+S3)	0.491	0.000	1.000	0.020	0.070	0.076
	Y4 (S4+S5)	0.000	0.704	0.398	1.000	0.938	0.153
	Y5 (S5+S6)	0.061	0.000	0.019	0.044	1.000	0.490
	Y6 (S4+S6)	0.016	0.014	0.000	1.000	0.031	0.310

The preliminary tests show that ANN can recognize appropriately chosen communication patterns and their combinations. It can recover the original signal patterns even under high noise suppression.

4.2 Simulation Results for Linear Pattern Envelope Construction Algorithm

The modulation method forms the basis of PBCS. The first study has an innovative approach about the shift keying on all three features of communication signal. According to the first approach, the enlargement of frequency bandwidth usage was the most important problem. Therefore, the band-pass filter is applied to the constructed signal patterns. Filtering process solved this problem. One example for the unfiltered signal patterns and their spectral distribution for 4-pattern (2-bit) glossary are located in Figure 4.6. The intervals for the features are selected as Amplitude (6-24V), Frequency (500-505 MHz) and Phase (0-360°).

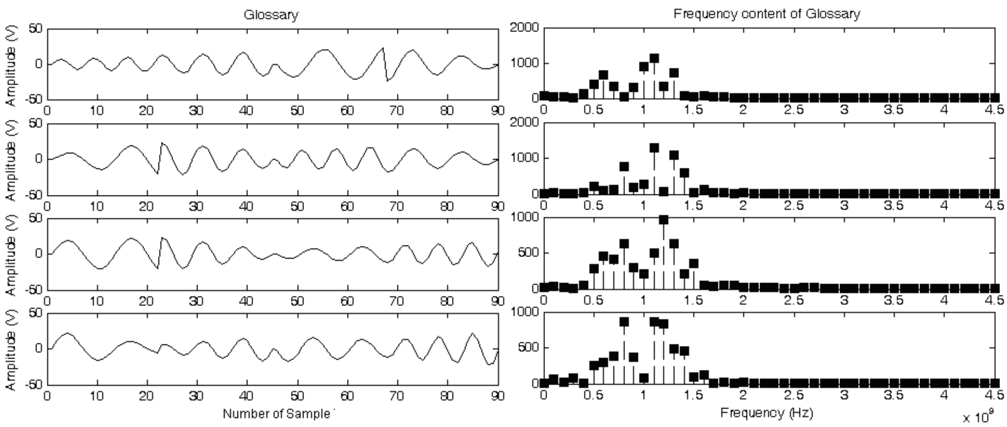

Figure 4.6 : AFPSK-based glossary and its spectral distribution.

The filtered signal patterns are shown in Figure 4.7. The spectral distribution is observed in satisfied interval. The performed tests until here and the results of them are published as a conference paper (Orcay and Ustundag, 2008).

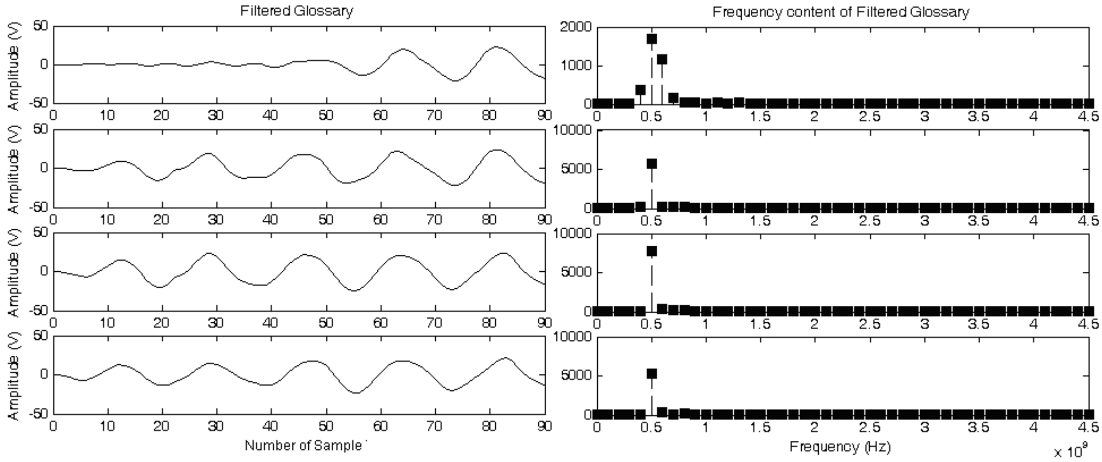

Figure 4.7 : The band-pass filtered glossary and its spectral response.

However, the continuity between the glossary patterns cannot be provided with this approach. This situation affects the performance of overall system. Due to fact, different construction algorithms for signal pattern were developed. First stage, three different Linear Pattern Envelope Construction (LPEC) methods is proposed for glossary construction. They are named as rotational, sequential and interval. The phase shift keying procedures for these methods was introduced in Figure 3.4. The performance of these methods is evaluated in this section.

The constructed glossaries by LPEC methods and their spectral distributions are shown in Figure 4.8. In this figure, the column at the left hand side shows the glossary patterns. Y-axis represents the amplitude of them. On the other hand, the column at the right hand side shows the FFT response of the related signal patterns in terms of power.

The spectral distribution of the glossaries is the most important criteria in terms of applicability for baseband application. Figure 4.8 shows the difference between them in terms of applicability. According to this figure, “Interval” method is the most applicable method for the signal pattern construction. In contrast to that, “Rotational” method is the worst case in LPEC algorithms. Since the change in phase is so fast in “Rotational” method, the result is obtained as expected figure.

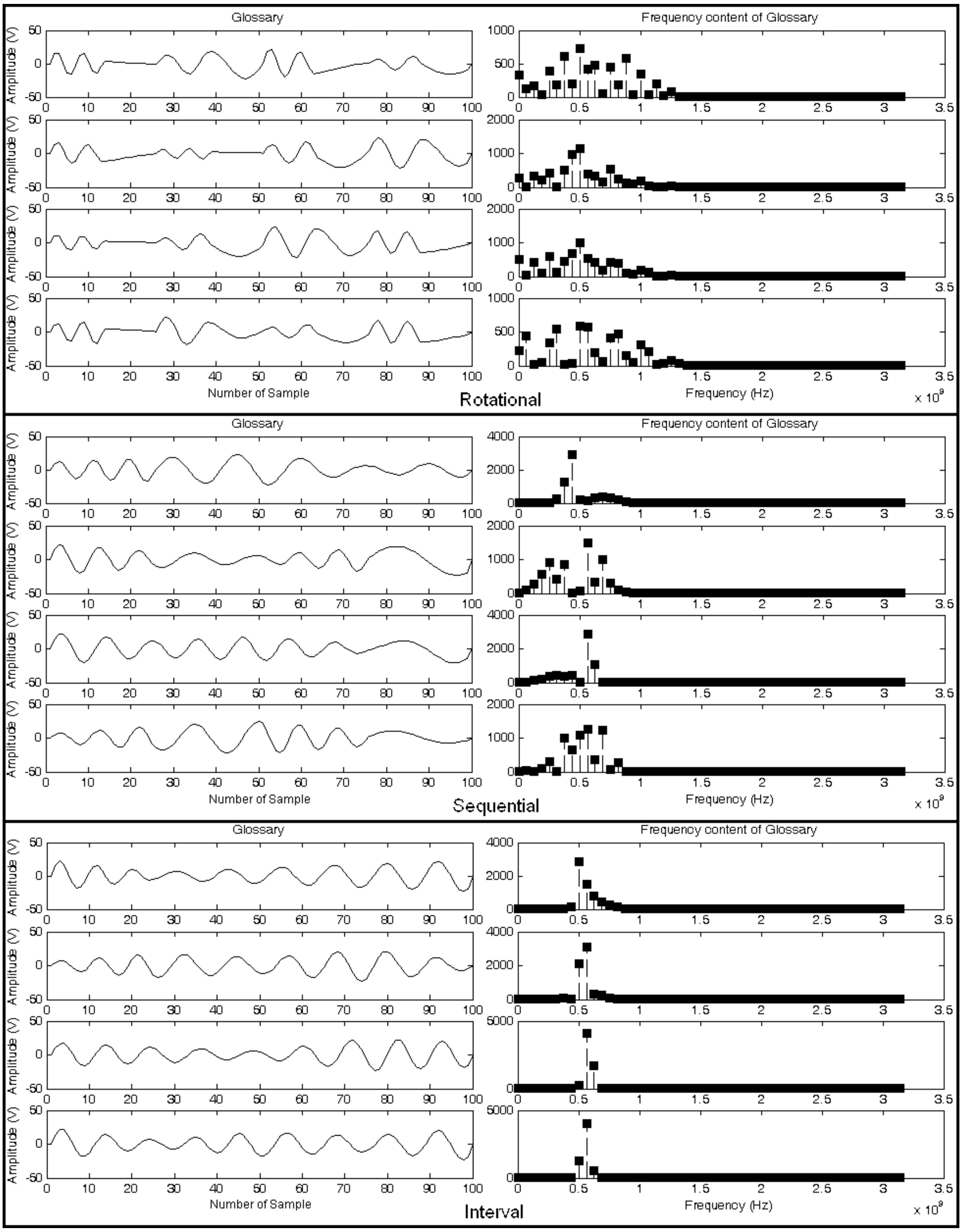

Figure 4.8 : LPEC algorithms (Rotational, Sequential and Interval).

The spectral bandwidth limit for baseband or spread spectrum communication is not satisfied by the usage of LPEC algorithms alone. These approaches can be used in Ultra Wideband application or they need to use a filtering process. Therefore, the band-pass filtering process is applied to all glossaries for baseband application. According to the result shown in Figure 4.9, the spectral bandwidth distribution of glossaries has better figure in terms of applicability.

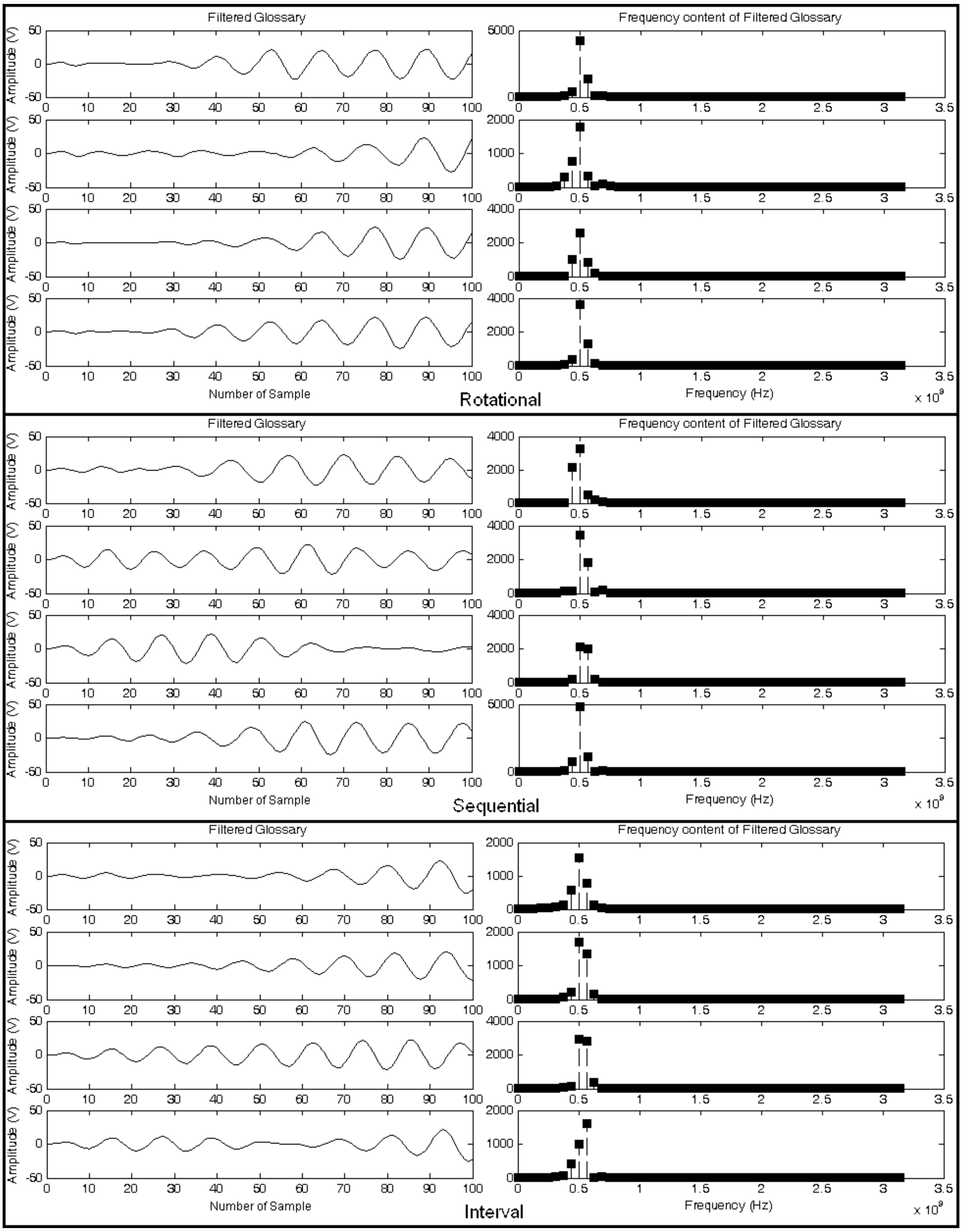

Figure 4.9 : LPEC-based filtered glossary and their FFT responses.

However, the signal patterns constructed by LPEC algorithms are not proper to use in the baseband or spread spectrum communication without band-pass filtering process. Due to its applicability, “Enhanced Interval” algorithm is developed to construct the glossaries. The signal patterns and the regarding spectral distribution is located in Figure 4.10.

According to the comparison between Figure 4.9 and Figure 4.10, the best choice in LPEC algorithms is “Enhanced Interval” method. However, millions of alternative patterns can be constructed by LPEC algorithm. The space should be limited and the most uncorrelated patterns in the space should be selected to increase the performance. The clustering and/or classification methods solve this problem. Vector Quantization (VQ) is selected for this purpose. The well-know software vector quantization tool, namely CLUTO (Url-1), is used to cluster the signal patterns. The distance criterion between the patterns is selected as the correlation coefficient. The training set of VQ is created with 1024 patterns. Each pattern in set is constructed by “Enhanced Interval” rules. CLUTO clusters this set regarding to the glossary bit level. For instance, it creates 4 centroids for 2-bit glossary. The closest pattern to the centroids is assigned as the glossary pattern.

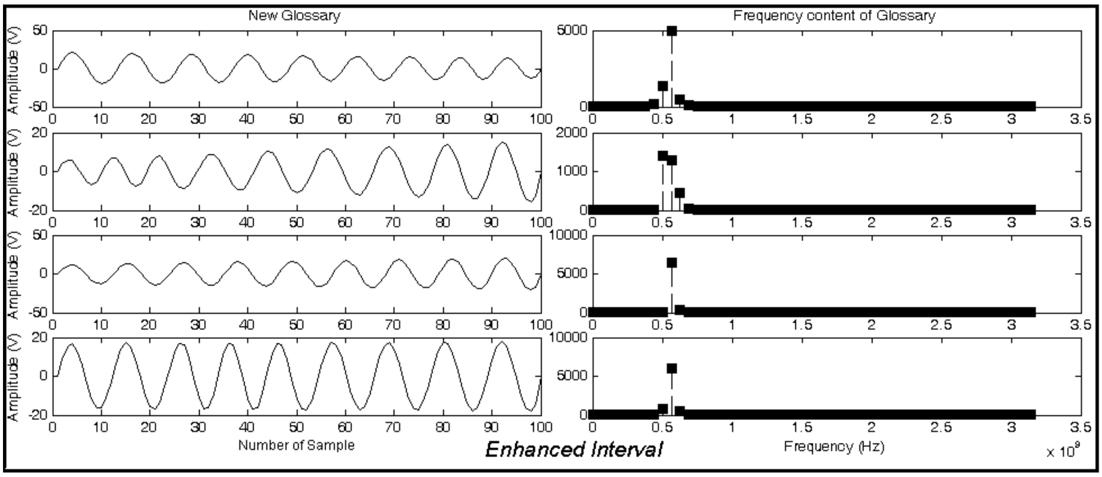

Figure 4.10 : The spectral distribution of enhanced interval technique.

CLUTO determines the most uncorrelated glossary patterns. These patterns are filtered with the band-pass filter. ANN is trained with these filtered glossary patterns. The tests are performed with AWGN channel model. The results are presented in BER vs. SNR space and the graph is given in Figure 4.11, where this system can work properly even at -18dB SNR value with 1-bit glossary. This means that the system is very robust against to noise effect with “Enhanced Interval” method of LPEC algorithm. However, the tests do not guarantee the continuity due to the band-pass filtering process.

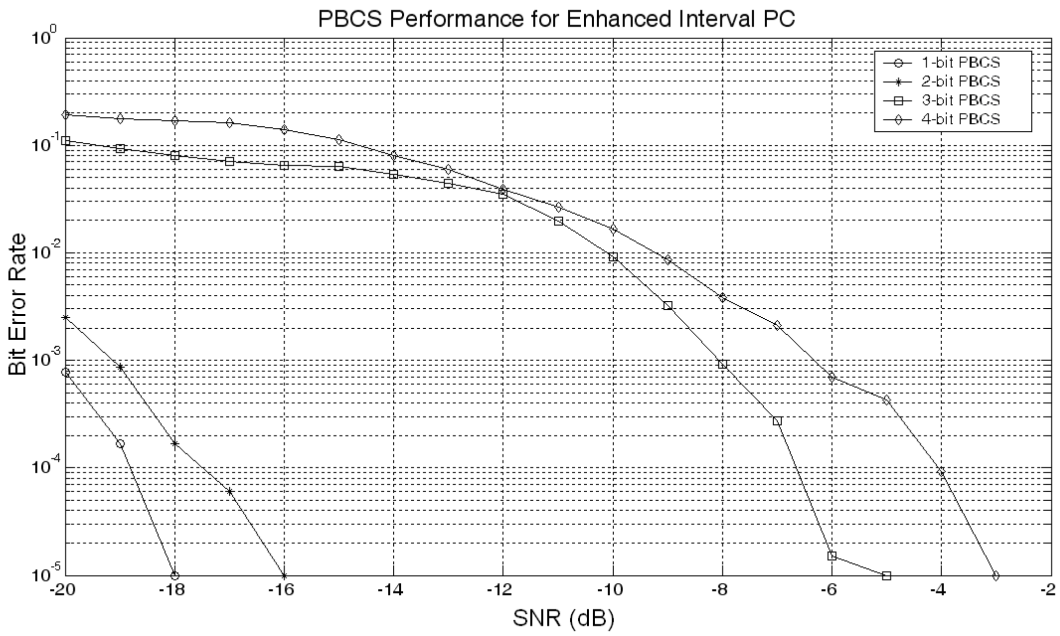

Figure 4.11 : LPEC performance for enhanced interval.

After the recognition capability of MLP is evaluated in realistic communication platform, it is compared with RBNN and WBNN to evaluate the performance change depends on the ANN type. These tests are performed for only 1-bit and 2-bits glossaries. According to graph in Figure 4.12, the best performance can be obtained from MLP for this problem. This result guides the direction of study in terms of ANN selection.

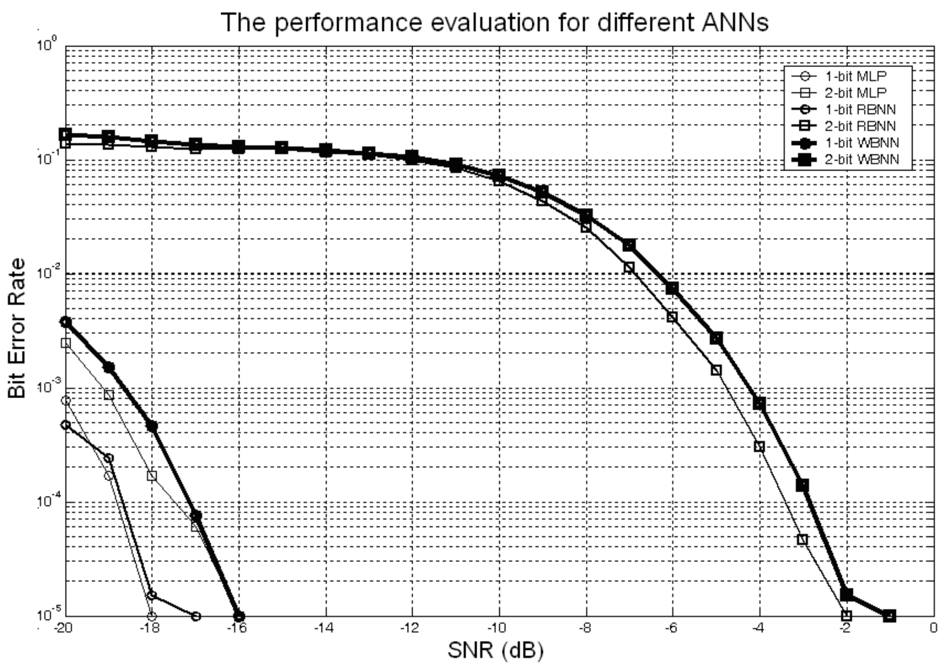

Figure 4.12 : The comparisons of MLP, RBNN and WBNN.

The problem here is that the glossary patterns are filtered alone. The performance evaluation for the concatenated set of patterns (test pattern) will affect the overall performance of the system. It mainly could not guarantee the continuity. Due to fact, the next studies are performed for extended test pattern. This concept will be explained at the next subsection.

4.3 Simulation Results for Sinusoidal Pattern Envelope Construction Algorithm

The glossary space in LPEC algorithm is huge and it has a nondeterministic structure. The alternative method of pattern envelope construction is developed to handle this handicap. This is named Sinusoidal Pattern Envelope Construction (SPEC) algorithm. The glossary space for this new approach is more deterministic than LPEC's one. SPEC space has two parameters. These are "Depth" and "Level". "Depth" determines the number of half sinus changes. "Level" identifies the number of intervals for each feature of signal pattern. The number of all the possible routes in any space was given in Figure 3.5, as an example. This example is generalized here.

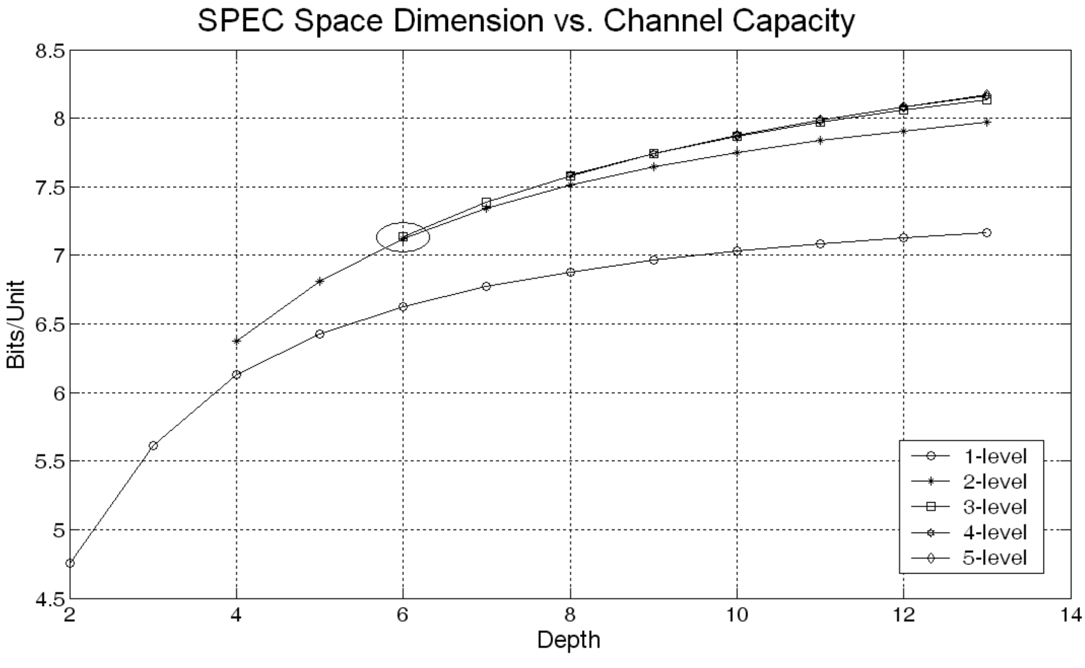

Figure 4.13 : SPEC space dimension vs. channel capacity.

The response of the system for different depth and levels are indicated in Figure 4.13. This figure is drawn with the following formula

$$\varsigma = \frac{\log_2 \rho^3}{\upsilon} \tag{4.1}$$

where, ρ shows the number of possible routes in the selected SPEC space. The route with the highest frequency determines the frequency of related SPEC space. This is shown as υ in units. The channel capacity (ζ) of any SPEC space is calculated in terms of bits/unit as shown in equation 4.1.

Each curve in Figure 4.13 represents the level for SPEC space. If the number of depth and/or level is increased, the channel capacity of the SPEC space is getting better. According to this figure, channel capacity goes to saturation, if these parameters are increased enough. However, the increment in the number of depth and level directly affects the operational complexity of the system. Due to fact, the closest point to the saturation with low complexity should be determined as a working point. Since the most satisfied point in this graph around Depth is 6 and Level is ±2 (circled in Figure 4.13), the dimension of SPEC space for the test platform is prepared with these values.

2-bit glossary is constructed by the usage of SPEC algorithm as an example. The spectral distribution of the glossary is given in Figure 4.14. According to this figure, the glossary constructed by the usage of SPEC has better shape than the previous studies.

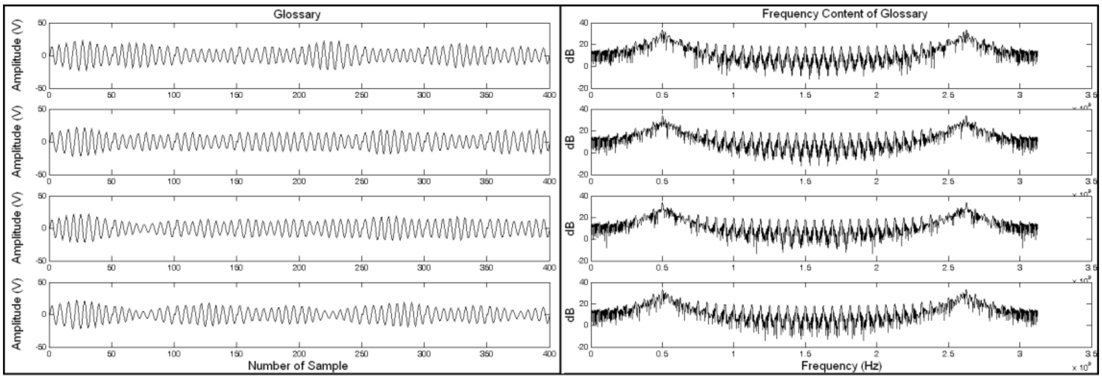

Figure 4.14 : SPEC-based glossary and its spectral response.

The signal patterns in 2-bit glossary are concatenated randomly. The obtained long signal pattern is named test pattern. The spectral distribution is measured with this pattern. The central frequency is selected as 500MHz. The interval for frequency is determined as 1MHz. The bandwidth need for the test pattern is calculated for 3dB loss from the peak point. The result is measured as 800kHz. This simulation is shown in Figure 4.15.

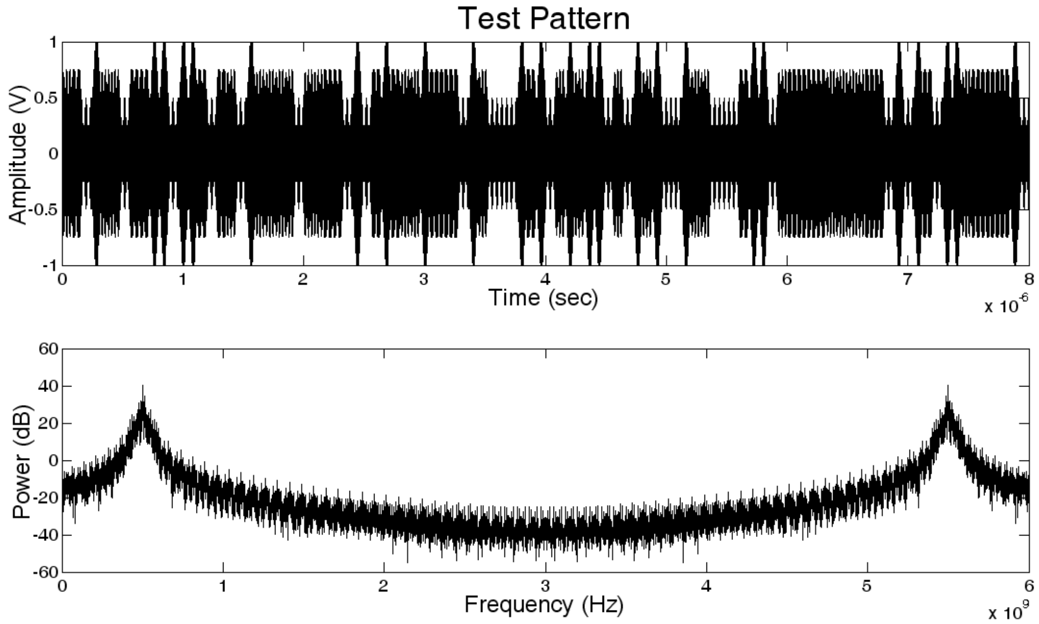

Figure 4.15 : The test pattern and its spectral distribution.

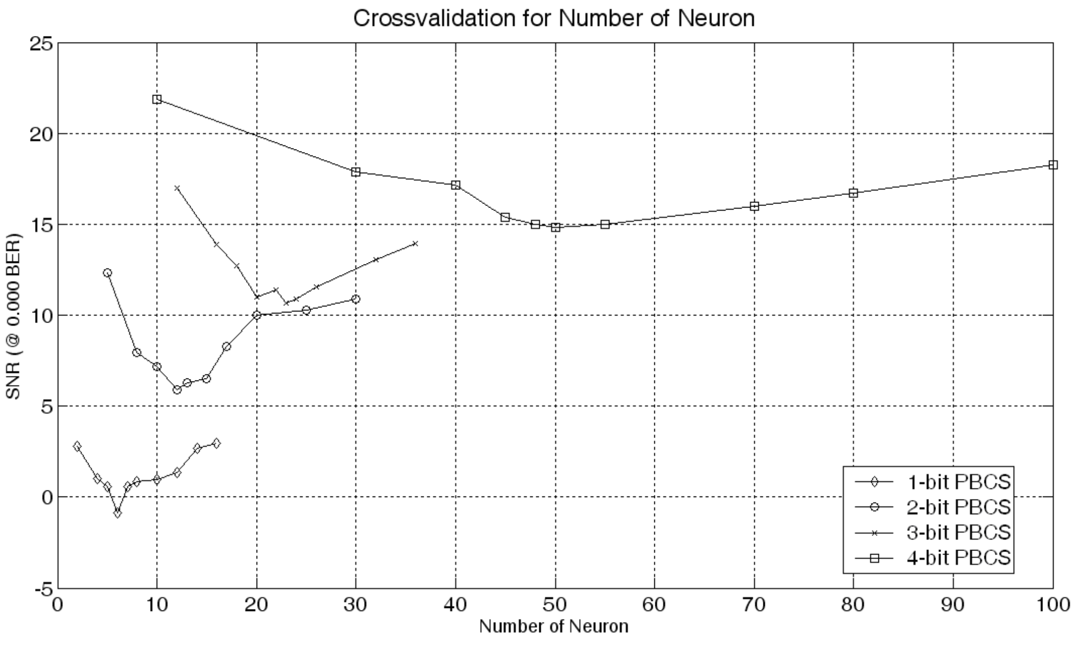

Figure 4.16 : Cross-validation implementation to LPEC glossary.

The performed tests showed that the best way to construct the glossaries is SPEC algorithm. The dimension of SPEC space is determined with accurate values. The main objective of this study is to design a flexible system. The proposed system should work on any part of SNR vs. LSE space depending on the user expectations.

In line with this objective, distinct glossaries are constructed for each bit level. MLP is also designed for each separately. The number of hidden layer, momentum coefficient and sigmoid coefficient are selected as constant. The learning rate is changed adaptively in training phase. Only the number of neuron is missing in the design of MLP. This process is handled by the usage of cross-validation. The results of this process for distinct glossaries are given in Figure 4.16 as an example.

Y-axis on Figure 4.16 shows the noise robustness of glossary in terms of SNR at approximately zero BER. Deep points in the graph represent the best number of neuron for the related glossary. The cross-validation operation is applied to all of the following tests. The glossaries and the uniquely designed MLPs for them are used to test the performance in SNR vs. BER space. The recognition capabilities of MLPs for the distorted glossaries by AWGN are given in Figure 4.17.

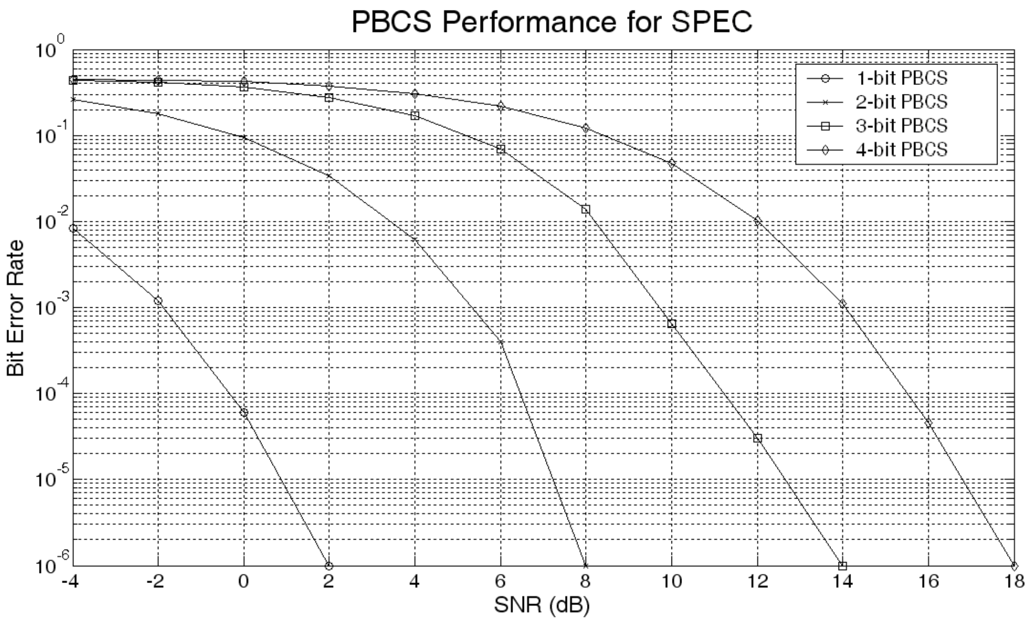

Figure 4.17 : The initial PBCS performance with SPEC algorithm.

The aim of Figure 4.18 is to indicate the recognition capability of PBC visually. The sample signal pattern in this figure shows any pattern in 1-bit glossary. This selected pattern is distorted with AWGN at 2dB and -2dB SNR values. These distorted forms are shown in the same figure respectively. According to Figure 4.17, ANN can recover the signal located at the bottom of Figure 4.18 with 0.125% BER. On the other hand, if the SNR value is increased to 2 dB, the success rate for the recognition performance dramatically increases. ANN can recover the distorted signal pattern located at the middle of Figure 4.18 with 10^{-6} BER.

According to simulation results, the parameters for glossary construction and ANN design are determined well for PBC.

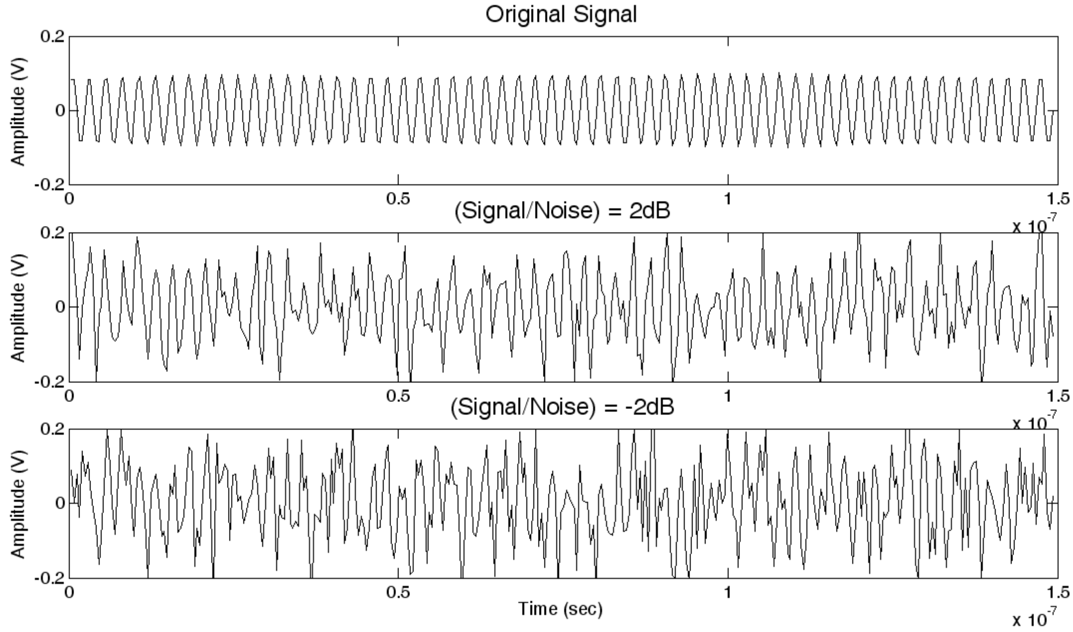

Figure 4.18 : The noise effect on the glossary pattern.

4.4 PBCS Behavior in Baseband Application

The proposed pattern construction method (SPEC) is used for baseband application. The aim of this study is to prove the applicability of the method for real-time applications. The semi-hardware test platform is designed as in Figure 4.19.

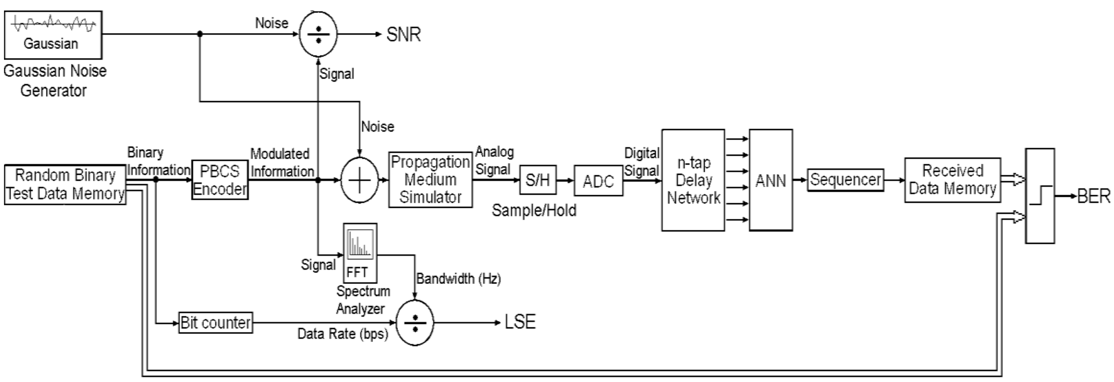

Figure 4.19 : Block diagram of test platform for PBCS.

The central frequency is selected 8 kHz and the spectral bandwidth utilization is calculated by the 3dB power loss. The spectrum analyzer measures the spectral bandwidth utilization. Each pattern is sampled by 40 kHz Analog to Digital Converter (ADC). The digital information is fed into ANN by the use of Delay Network. ANN decodes the digital signal to binary words and the sequencer concatenates the output of ANN. The comparator compares the original and reconstructed signal. BER is derived from the end of this process. According to test results in semi-hardware platform at 8kHz central frequency, PBCS can recover the original data at receiver side for different LSE and SNR values. These results are presented in Table 4.4.

Table 4.4: The results for Test Platform at 8 kHz.

Glossary	Input Bandwidth (Hz)	Output BER	SNR (dB)	LSE (bps/Hz)
G_I	5.75	10^{-5}	20.25	10.869
G_{II}	7.21	10^{-5}	15.75	8.668
G_{III}	381.56	10^{-5}	-3.10	0.163

According to the test platform results, the duration of decoding process takes 0.20 sec for the test sequence with duration of 1 sec. It proves that PBC can be implemented with this structure, if the hardware platform is designed for proper parameters. These results are published as a journal paper (Ustundag and Orcay, 2011).

4.5 Improvement LSE via Manageable SNR

The most important innovation in PBC is to design a system, which can manage SNR depends on the user request in terms of LSE. According to the user request and channel conditions, it switches the communication signals and find out the best point. 35 different glossaries are constructed with different boundary conditions for three signal features (A, F and P). These glossaries are constructed by the usage of SPEC algorithm for depth 8 and level ±3. The central frequency is selected as 1 kHz and the sampling rate is defined as 24 kHz. The sampling process is performed by the utilization of 8-bit ADC. The bandwidth is measured with 3dB loss from the peak and the average bandwidth measured as 50 Hz with this measurement approach. The mean value of the peak-to-average power ratio (PAPR) for whole dataset is calculated as 7.1712 dB with the standard deviation 0.2141. The guard interval is selected as 0.5 for these tests. These 35 glossaries are located in LSE vs. SNR space as in Figure 4.20.

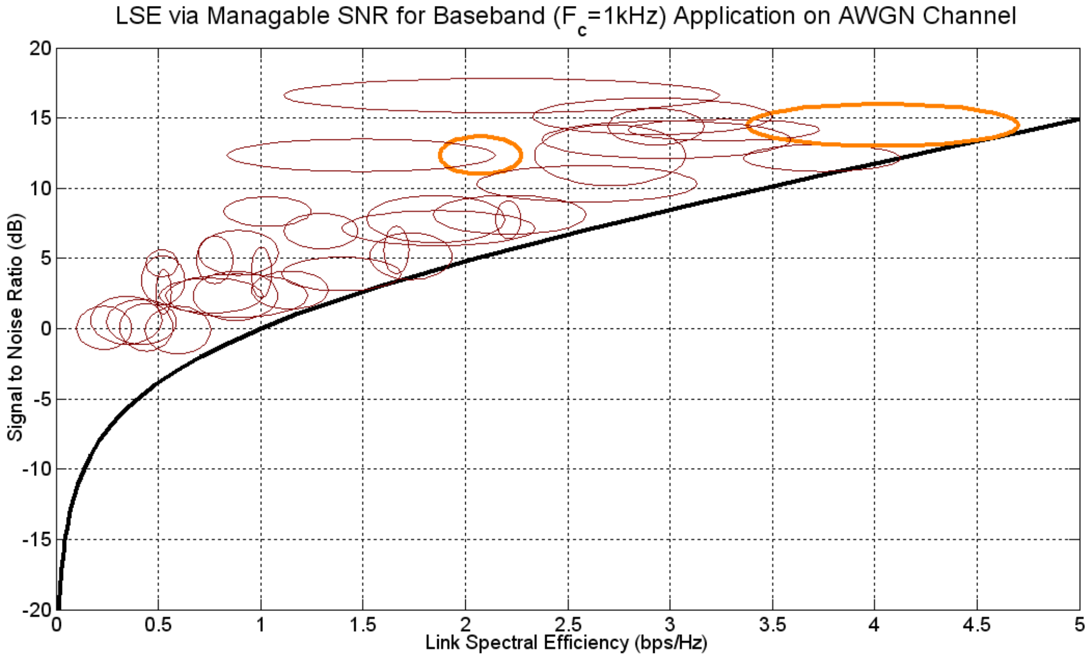

Figure 4.20 : LSE vs. SNR for baseband application in AWGN channel.

The Shannon boundary (bold curve in Figure 4.20) is drawn by the usage of Shannon limit theorem. The equation 4.2 is derived from equation 1.1. Just the parameters of equation 1.1 are adapted to the space parameters of SNR vs. LSE.

$$\frac{C}{B} = \mathrm{LSE} = \log_2\left(1+10^{\left(SNR/10\right)}\right) \qquad \textbf{(4.2)}$$

Figure 4.20 shows the flexible structure of PBC in SNR vs. LSE space. Many distinct glossaries can be constructed by the utilization of SPEC algorithm. According to the user request and channel conditions, PBCS switches to the best point for optimizing the communication system performance. The ellipses in Figure 4.20 show the working area of regarding glossary at 10^{-5} BER. The most significant feature of the proposed method is its competence to operate in any location above the Shannon boundary based on the user request. This result provides the manageable SNR capability for different LSE values. From adaptive modulation perspective, this feature also allows the user with the flexible and efficient utilization of the spectral bandwidth. In fact, PBC is an adaptive modulation technique that proper to use in CR with the extension of control unit to cognitive engine.

On the other hand, ANN design for each glossary is another important point for the system performance. MLP is designed with single hidden layer for each glossary. The number of neurons in hidden layer is determined with the cross-validation method. One example to define the number of hidden layer neurons is given in Figure 4.21. This algorithm works in recursive structure. When we compared with the previous cross-validation approach with this one, the new one scans the wider space to define the best performance for the related MLP. The momentum and sigmoid coefficients are set to 0.5 and 0.6 respectively. The learning rate is adjusted in adaptive manner.

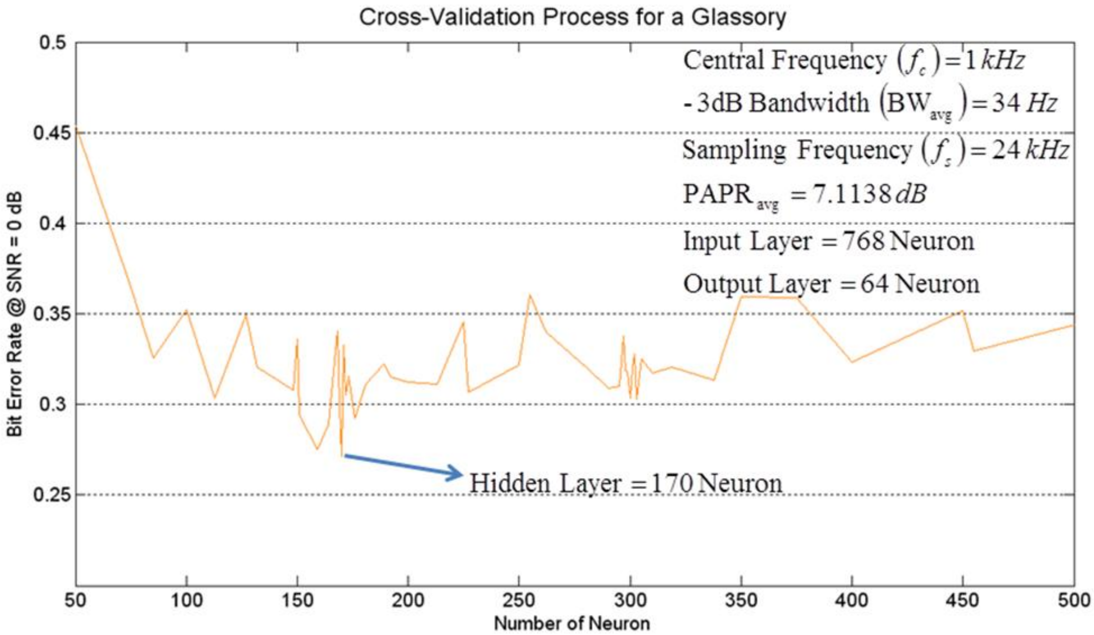

Figure 4.21 : Cross-validation for SPEC-based glossary.

The utilized glossary in Figure 4.21 is the most complex one in the glossary space in terms of the number of neuron in hidden layer. Therefore, this could be used as an example to calculate the real-time applicability of PBC in hardware platform. This most complex MLP has 768 neurons in input layer, 170 neurons in hidden layer and 64 neurons (6-bit glossary) in output layer. It means that the hardware platform should support 130.560 (=768x170) multiplications in one loop. The FPGA is selected to perform this calculation. One of the high performance FPGA, named as Altera Stratix V SGSD8, can handle this huge amount of calculation. The clock frequency of this device is 250 Mhz and it can perform over 10K multiplication for 9-bit information in one loop. It means that the system needs approximately 13 loops to handle this process. It takes 52ns. On the other hand, the signal pattern duration is 13ms for 1 kHz central frequency and 32 period. This calculation proves that the one of the most complex MLP to recognize any glossary can be implemented in hardware platform.

The hardware platform with evaluation kits is under construction. The researchers are designing this platform with the utilization of DSP at transmitter side. Since the parallel input need at ANN, the designers selected FPGA at the receiver side. This work is defined as a future works. The following baseband application results are obtained by the utilization of two computers at 1kHz central frequency. In the initial phase, the comparison between PBC and non-adaptive conventional methods is located at the following sub-section.

4.6 Performance Comparison between PBC and Non-adaptive Conventional Methods in AWGN Channel

The other most important aim of PBC is to minimize the area under the graph for E_b/N_0 vs. BER space. The details of this optimization were explained in section 2. The noise robustness performance of PBC and the position of it against to Shannon boundary is discussed in this section. Moreover, the performance comparison with MF and WM is given in separate evaluations.

Figure 4.22 shows the position of PBC against to Shannon boundary. The signed part of Figure 4.22 with dotted line represents the Shannon Limit. This limit is derived from the ultimate Shannon limit calculation, which is explained systematically as below. The equation 4.3 uses equation 1.1 and derives the different form of this equation.

$$
\begin{aligned}
&C < B\log_2\left(1+\frac{S}{N}\right)\\
&\text{LSE} = \frac{C}{B} = 2R_l < \log_2\left(1+2R_l\frac{E_b}{N_0}\right)\\
&\frac{E_b}{N_0} > \frac{2^{2R_l}-1}{2R_l}
\end{aligned}
\qquad \textbf{(4.3)}
$$

The Shannon limit is defined as r goes to 2 in the literature. On the other hand, the ultimate Shannon limit defines as "r" (= $2R_l$) goes to zero (r→0). If "r" goes to zero, the limit of the equation mentioned above goes to zero. Therefore, the solution can be calculated by the utilization of L'Hospital rule. This calculation is given in equation 4.4.

$$\text{if} \lim_{x\to 0}\frac{f(x)}{g(x)} = 0 \quad \Rightarrow \quad \lim_{x\to 0}\frac{f(x)}{g(x)} = \lim_{x\to 0}\frac{f'(x)}{g'(x)}$$

$$\lim_{x\to 0}\frac{2^x - 1}{x} = \lim_{x\to 0}\frac{2^x \ln(2)}{1} = \ln(2) = 0.69 \qquad \textbf{(4.4)}$$

$$10\log_{10}(0.69) = -1.59\,\text{dB}$$

According to this calculation, the ultimate Shannon limit equals to -1.59 dB in E_b/N_0 vs. BER space as mentioned in Figure 4.22.

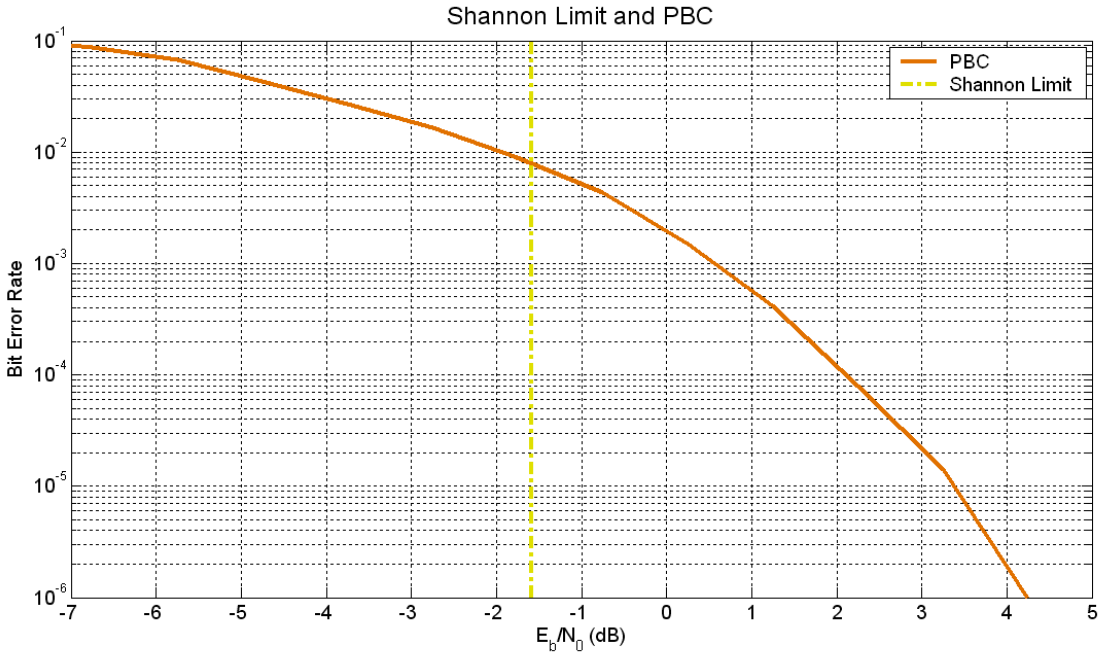

Figure 4.22 : The position of PBC against to ultimate Shonnon limit.

Figure 4.22 shows that one of the best performances for the glossaries modulated by PBC can work approximately 0.0075 BER at the ultimate Shannon limit. Although this BER cannot be acceptable for any communication platform, this value is at very good position, when it is compared with the existing modulation techniques.

The performance of PBC is first compared with non-adaptive supporting and modulation techniques in AWGN channel. The comparison between PBC and the well-known supporting technique MF is given in Figure 4.23. This evaluation is performed to discuss the performance of PBC in terms of noise robustness. One of the 1-bit glossaries from the glossary space represented in Figure 4.20 selected for this comparison. The LSE value of selected glossary is 0.5181 bps/Hz and it uses 41 Hz bandwidth in the communication medium. The PAPR of this glossary is 7.0062.

According to the results, the selected glossary constructed by PBC modulation has 2 dB better performances at 10^{-5} BER against to MF supported BPSK.

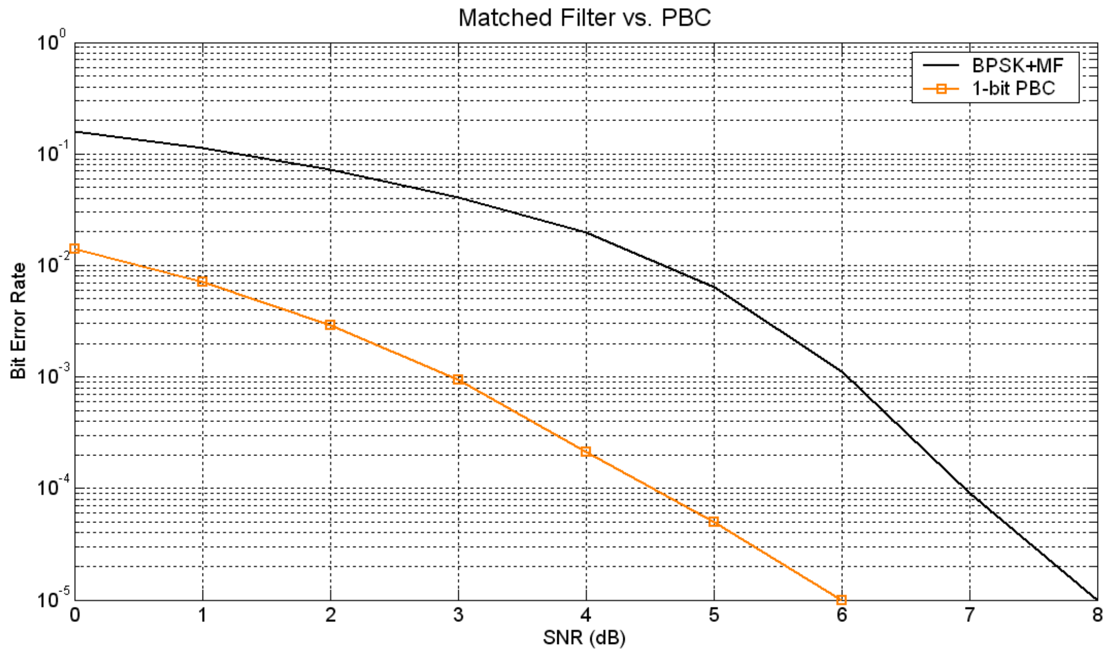

Figure 4.23 : Matched Filter vs. PBC in AWGN channel.

This comparison between PBC and MF was performed one of the previous works. The results were published as a journal paper (Orcay and Ustundag, 2010). The given results in this journal are evaluated in the different test parameters and they are given in this thesis with the new shape.

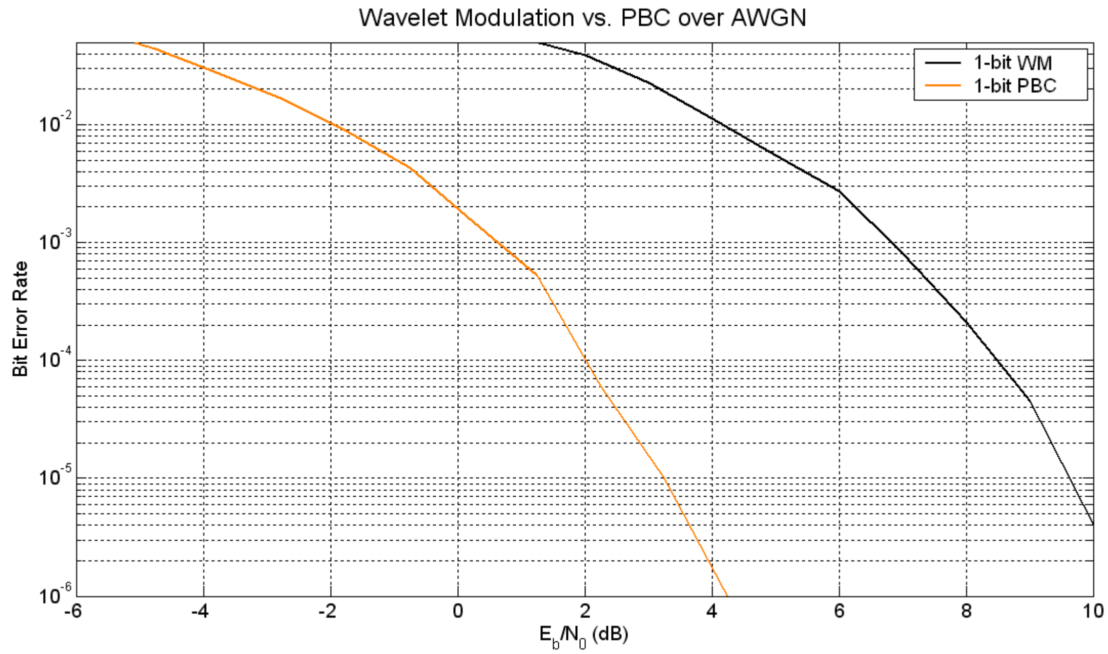

Figure 4.24 : Wavelet Modulation vs. PBC in AWGN channel.

Another comparison is performed between PBC and WM in AWGN channel for non-adaptive manner. The same glossary mentioned in Figure 4.24 is used for this comparison. The evaluation parameters are changed. This comparison is given in E_b/N_0 vs. BER instead of SNR vs. BER space. According to the results, PBC has around 6 dB better noise robustness then MW in terms of E_b/N_0 value.

4.7 Performance Comparison between PBC and Adaptive Conventional Method QAM in AWGN Channel

The above results point out the non-adaptive data rate performance for PBC and the comparison with some related conventional methods in terms of the noise robustness. The adaptive performance of PBC is evaluated in this subsection. PBC is compared with well-known adaptive modulation approach QAM. The performance of these two techniques are discussed in AWGN channel. One of the simulation results is given in Figure 4.25.

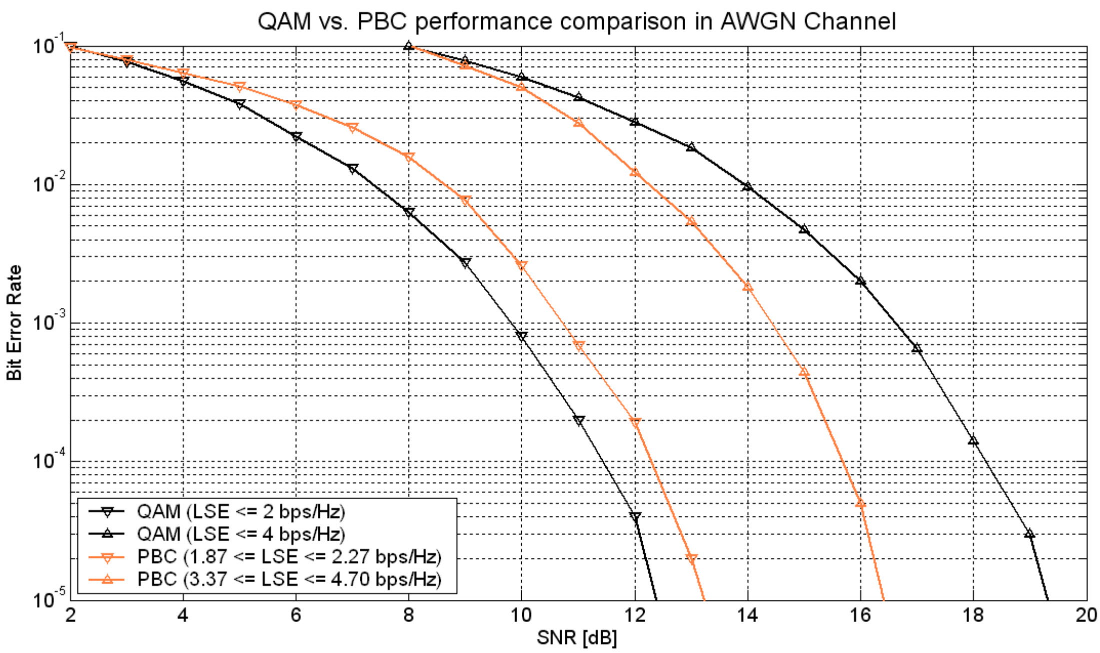

Figure 4.25 : QAM vs. PBC: Non-adaptive performance (Scenerio 1).

The glossaries in this example need to use 36.71 Hz bandwidth for performing the communication and the average PAPR for this communication is 7.6893 dB. LSE values for the selected glossaries have 2 bps/Hz and 4 bps/Hz in average (Scenerio 1). The results in this example shows that PBC modulated information has better performance at higher bit rate. These glossaries are used for the adaptive scenerio and the result is given Figure 4.26.

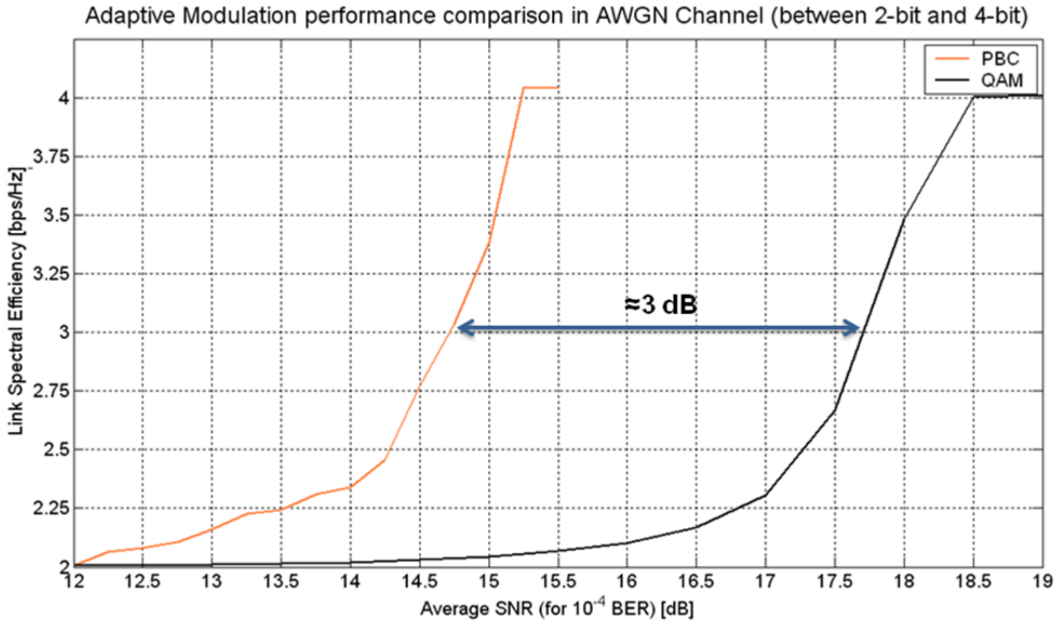

Figure 4.26 : PBC vs. QAM in AWGN channel (for Scenerio 1).

According to these simulation results, PBC modulated infromation with the defined glossaries has 3 dB better pefromance than QAM in average. Due to the better performance of PBC for higher bit rate, the difference between two modulation approach is increasing for higher bit rate.

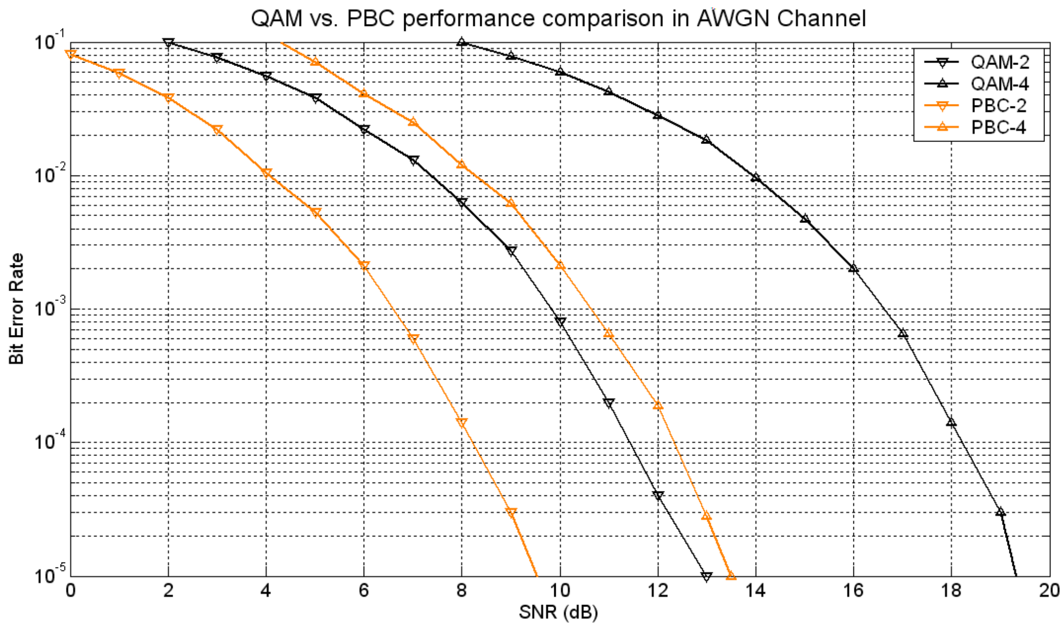

Figure 4.27 : QAM vs. PBC: Non-adaptive performance (Scenerio 2).

Another test is performed for another two glossaries in the glossary space. These two glossaries have maximum LSE values as 2 bps/Hz and 4 bps/Hz (Scenerio 2). This comparison is more fair than the previous one. Because, QAM-2 and QAM-4 can transmit maximum 2 bps/Hz and 4 bps/Hz sequentially. The non-adaptive performance of the seleceted glossaries against to QAM equvalent is given in Figure 4.27. The adaptive performance of these glossaries and the comparion with QAM for 10^{-4} BER is shown in Figure 4.28.

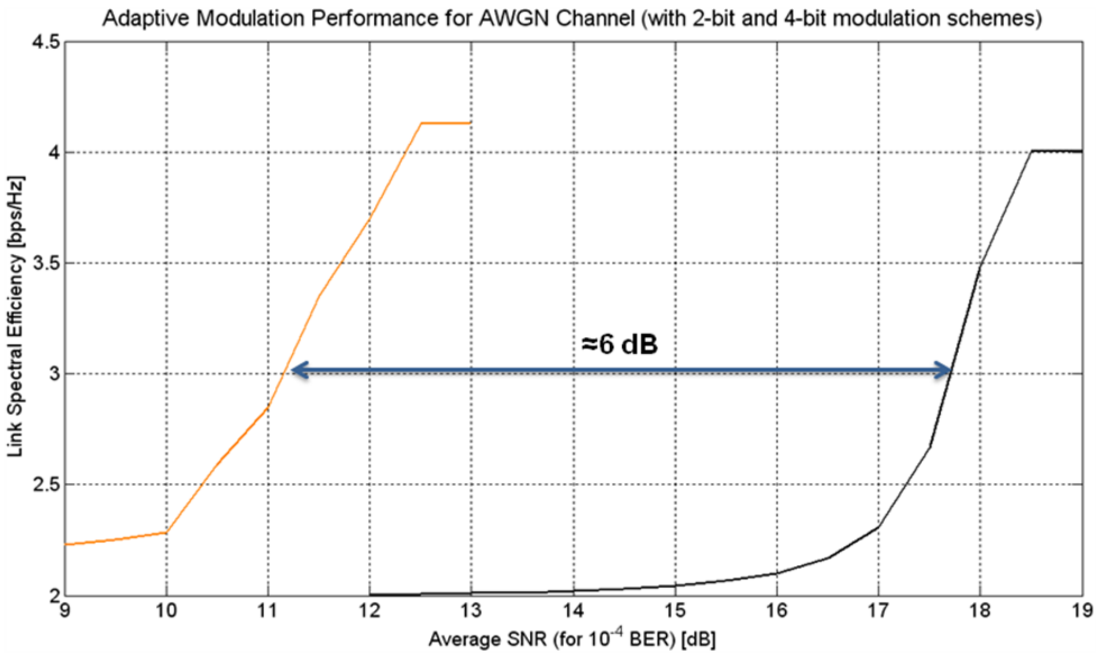

Figure 4.28 : PBC vs.QAM in AWGN Channel (for Scenerio 2).

According to this comparison, PBC can work on the communication channel with 6 dB less SNR value than QAM for the same LSE. This comparison also shows that PBC modulated information has better noise robustness than the conventional methods.

4.8 Multi-user Performance of PBC in AWGN Channel

According to above results, PBC performance is better than some related conventional methods in fixed or adaptive structures. Especially, PBC has more immunity against to high noise suppression. This part focuses on the other feature of PBC. The multi-user utilization ability of PBS has already shown in the beginning of this part with some basic tests. The performance of 2-bit glossary in the utilization of single user and multi user will be discussed in this subsection.

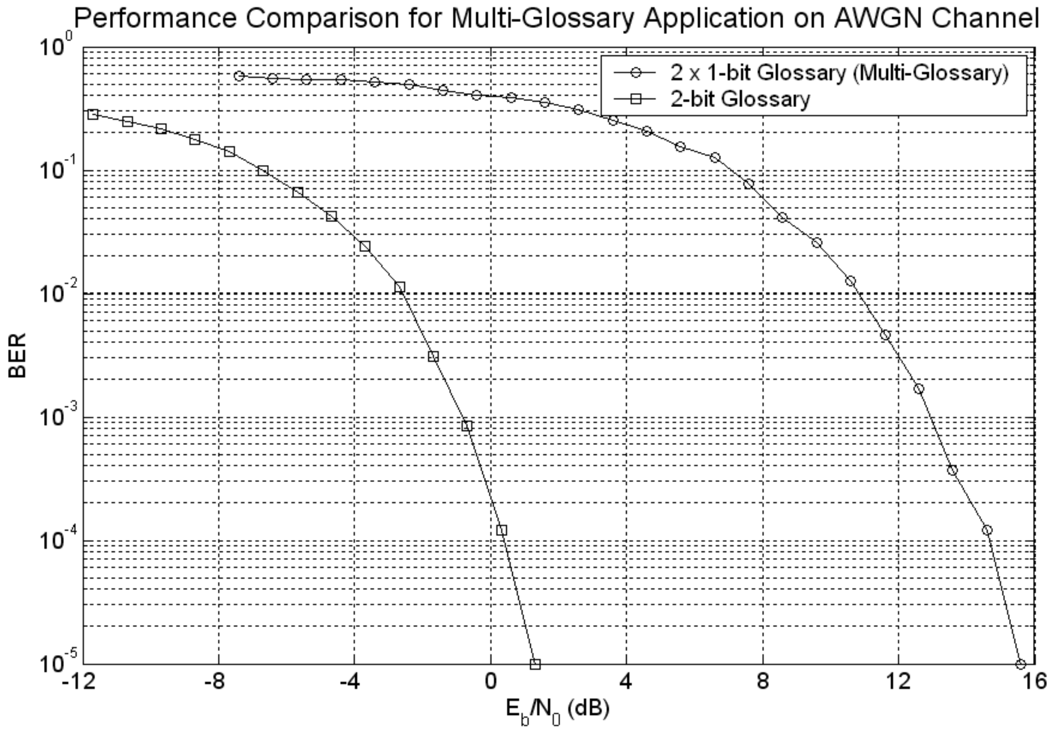

Figure 4.29 : Multi-user performance of PBC.

2-bit glossary is constructed by SPEC approach in Figure 4.29. This glossary is used for single user and the success rate obtained as 10^{-5} BER at around 2 dB E_b/N_0 value. Then, this glossary is divided into two parts and it is used for two users as 1-bit glossary. These patterns are transmitted through the communication channel simultaneously. According to semi-hardware platform test results, two users can transmit their own information under approximately 16 dB E_b/N_0 value. These results show that PBC can be used in multi-user systems but the performance reduces dramatically in terms of noise robustness as expected. The rest of simulation results are focused on IEEE 802.11g to show the expected performance of PBC for wide band and RF applications. These are out of scope of this thesis but located for defining future works to extend PBC modulation scheme.

4.9 Simulation Test Results for Future Works

Some tests are already performed for defining the future works. The results of these tests are located in this sub-section. The first test is related to the well-known communication band IEEE 802.11g. The filtering effect is given and the performance of PBC is evaluated in the first sub-section. Then, the performance comparison is performed between PBC and well-known modulation techniques, which are used in

CR applications. These are namely OFDM and PSM. The comparison is given in AWGN channel, due to the scope of this thesis. At the end of this sub-section, the PBC performance is evaluated in fading channel for the future works. The target of this thesis is the evaluation of the proposed modulation and coding technique in AWGN channel. However, the performance of any modulation technique in fading channel is very important to prove the applicability of the modulation approach in any communication channel. Therefore, some tests are performed on the simulation tool and the results are given at the end of this sub-section.

A glossary is constructed by SPEC algorithm to perform the filtering effect. Indeed, this glossary satisfies the bandwidth limitation for spread spectrum communication. However, it should be filtered to fit the channel conditions for IEEE 802.111g. Therefore, the spectral bandwidth utilization is simulated by the usage of two different masks. The simple filtering and the usage of IEEE 802.11g mask (IEEE Computer Society, 2008) filtering are given in Figure 4.30 and Figure 4.31 sequentially.

The comparison between the performances of the specified glossary usage with different masks is presented in Figure 4.32. The best performance is obtained to use the system without filter as expected. If the IEEE 802.11g mask is used to the test pattern, the performance is getting worse. The difference between the best and the worst performances is around 4 dB.

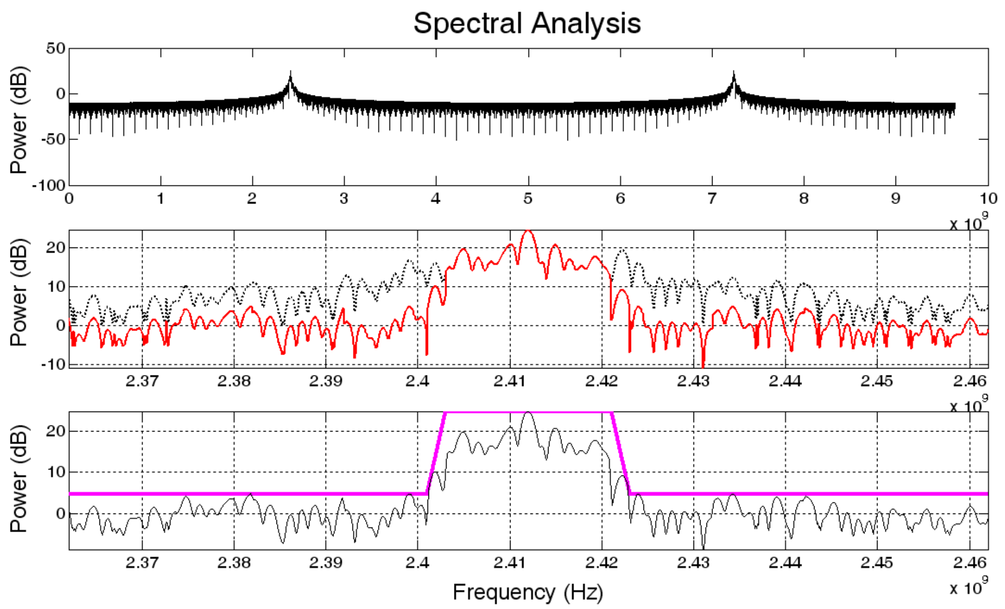

Figure 4.30 : The filtering test pattern with simple filter.

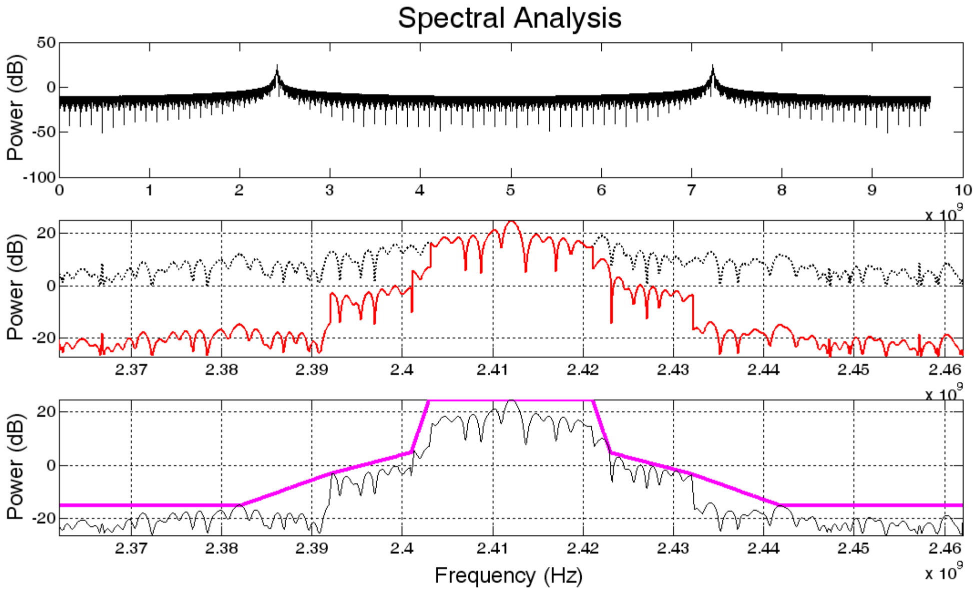

Figure 4.31 : The filtering test pattern with IEEE 802.11g mask filter.

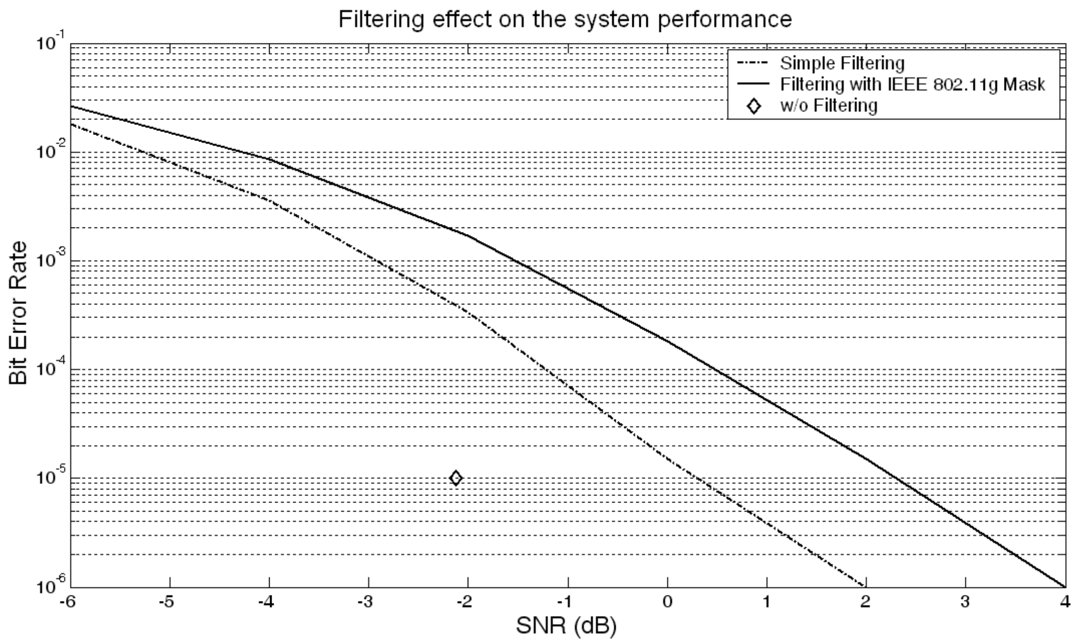

Figure 4.32 : Filtering effect on the system performance.

4.9.1 Performance Comparison between PBC vs. OFDM & IR-UWB (PSM)

OFDM is one of the popular techniques for CR applications. It has the flexible and manageable structure. Due to fact, it is proper to use in CR as mentioned before. OFDM uses the conventional modulation and coding techniques. It manages them according to the real-time spectral conditions. QPSK and QAM are the most popular modulation techniques used with OFDM. Therefore, the performance comparison between 2-bit PBC and QPSK-based OFDM is performed. The values for OFDM are obtained under IEEE 802.11.a/g standards. This means that the tests are performed for spread spectrum conditions (Kumar and Sharma, 2010). On the other hand, the glossaries for PBC is constructed for the central frequency 2412 MHz with 68 (sampling frequency is 24 times of central frequency) and 73 MHz (sampling frequency is 4 times of central frequency) bandwidth utilization. PAPR for these two glossaries are 7.1645 dB and 6.1066 dB sequentially. LSE values for them are 1.4913 bps/Hz and 1.379 bps/Hz in sequence.

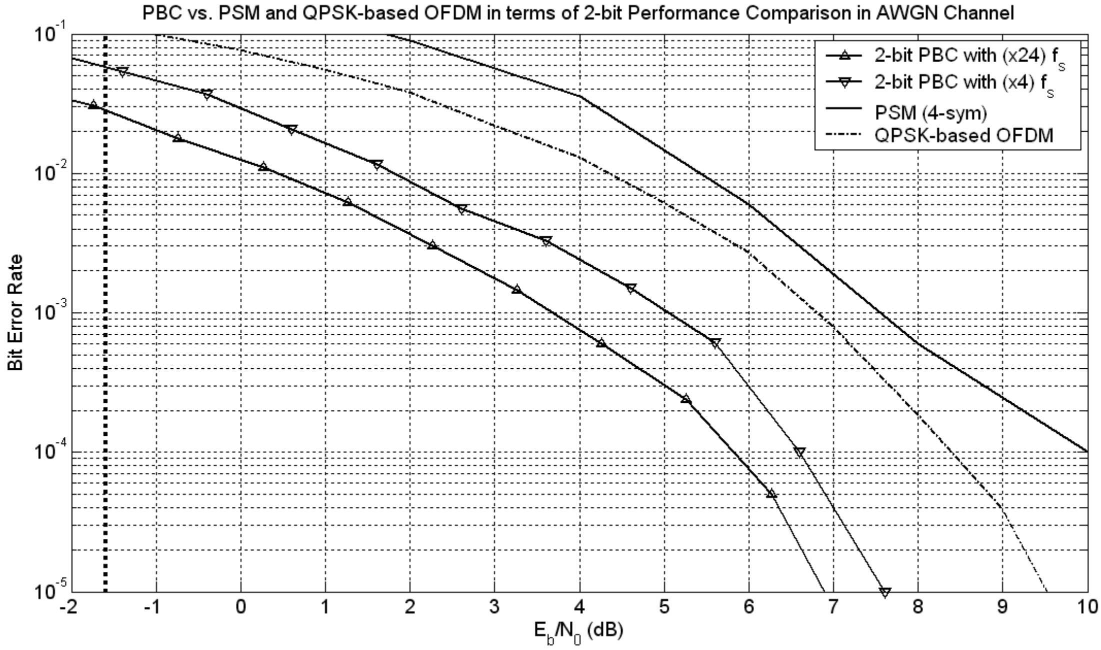

Figure 4.33 : PBCS vs. OFDM-based QPSK and PSM.

Moreover, IR-UWB is a different approach from narrowband or spread spectrum communication. PSM is taken as a reference for the comparison. Gaussian pulse is used as a base pulse and 4 of 8 different pulse shapes are used to perform the tests. The values for PSM are taken from the Ghavami's work (Ghavami et al., 2002). This structure encodes 2-bit information. Thereof, the results are compared with 2-bit PBC as OFDM case. The comparison between these two well known techniques and PBC is given in Figure 4.33.

According to tests results in Figure 4.33, PBC modulated information has better performance than these conventional methods for any BER. For instance, if the system works under 10^{-4} BER, PBC can work on 2.5 dB and 4.20 dB less than QPSK-based OFDM and PSM sequentially. This comment is valid for 24 times sampling rate of the central frequency. However, if the central frequency is very high like PSM and/or OFDM applications, this sampling rate cannot be applicable. Therefore, another test result is also given in the same graph. According to the lower sampling rate PBC can work on 1.5 dB and 3.20 dB less than QPSK-based OFDM and PSM sequentially. Although the sampling rate reduced 6 times, the noise robustness in terms of E_b/N_0 values is only 1 dB observed. The reason of this situation is mainly coming from the better LSE of PBC.

4.9.2 Performance Analysis for Multipath Model

The time delay estimation is one of the most important problems in communication lines. Many different methods are proposed in the literature to handle this problem. The hardware design for PBC has the most important problem for multipath resolving as usual. Multipath model decreases the performance of modulation techniques at different communication medium. The design for RF communication medium affects from this problem at the highest level. Therefore, solution advices were given in section 2. Since PBC is proper to use in time-invariant and AWGN channel model, any of channel estimation model in Figure 2.7 can be used to handle this problem. However, another solution advice is proposed in this thesis to solve this problem inspired by the utilization of transmitting space signal. According to Figure 2.9, if the pattern is constructed long enough, ANN can recover the original information. The following tests are performed to prove this claim.

Two different simulation test platforms are designed for determining the time delay resolving performance of PBCS. The test platform is designed by the ITU-R M.1225 (ITU-R M.1225, 2000) standard. The outdoor to indoor and pedestrian test environment tapped-delay-line parameters are presented in Table 4.5.

Table 4.5: Pedestrian test environment tapped-delay-line parameters.

Channel A			
TAP	Relative Delay (ns)	Average Power (dB)	Doppler Spectrum
1	0	0	Classic
2	110	-9.7	Classic
3	190	-19.2	Classic
4	410	-22.8	Classic

At the first stage, the base pattern duration (t_{data} in Figure 2.9) is selected longer than time delay for last tap (ΔT). The test results are given in Table 4.6. According to these results, if the reflected part of the signal comes as early as possible, the recovering performance of ANN decreases. When the base pattern duration goes to the arriving time of the last part of multipath, it goes to saturation and close to same level.

Table 4.6: The simulation results for Multipath Model.

	Input			Output		
Glossary	Duration Data(ns)	Duration Data + Space (ns)	Bandwidth (MHz)	BER	SNR (dB)	LSE (bps/Hz)
G_A	246	492	40	10^{-5}	16.0	0.4020
G_B	295	590	27	10^{-5}	17.5	0.4981
G_C	328	656	4.0	10^{-5}	23.0	3.0150
G_D	377	736	1.1	10^{-5}	22.5	9.5336
G_E	492	984	20	10^{-5}	18.0	0.4020

At the second stage, the beginning point of last tap is selected lower than the duration of the base pattern. The recognition performance of ANN increases, if it compares with the first stage case (G_E in Table 4.6). Since the tail of last tap did not effect to the second signal pattern, the performance of system increases as expected.

5. CONCLUSION AND DISCUSSIONS

This thesis proposes flexible and adaptive waveform construction algorithms. The defined approach in this work is named as Pattern Based Communication (PBC). The objective of conceptual system, Pattern Based Communication System (PBCS) is to improve the data transmission performance under communication channel limitations. It aims to develop a new modulation and coding technique. PBCS either optimizes the Link Spectral Efficiency (LSE) via manageable Signal to Noise Ratio (SNR) value based on user expectation or provides its user with the better noise immunity.

In this study, the PBCS transmitter matches the constructed signal patterns (waveforms) with the binary information. The pre-trained Artificial Neural Network (ANN) with the specified patterns (waveforms) at the receiver side recovers the distorted signal in the communication medium. The bit sequencer then concatenates the output of ANN for obtaining the original information.

The main goal of PBCS design is to construct a communication signal waveform, which works well under high noise suppression, namely Additive White Gaussian Noise (AWGN), on the communication medium. The simulation tests are focused on the system performance measurement under different noise levels.

First, the recognition capability of ANN is measured. The result showed that ANN could be used for the recognition of communication signal pattern. After that, the system design is taken into consideration. There are two main design parts in PBCS. The glossary (waveform) construction method, namely PBC, is the crucial part of the transmitter in terms of the efficiency. On the other side, ANN design is very important part for whole system performance. Many different glossary construction methods are proposed to find the best solution. According to the simulation tests, Sinusoidal Pattern Envelope Construction (SPEC) algorithm is determined as the best solution for baseband and spread spectrum communication. Moreover, Linear Pattern Envelope Construction (LPEC) and/or SPEC algorithms can be used for Ultra Wideband (UWB) communication. The important hint is that SPEC provides more deterministic way to construct signal patterns (waveforms). After the glossary construction method and ANN type are chosen, the design details are determined with the experimental tests. Many different tests are then performed to show the advantages of the proposed system against to the conventional methods.

One part of this study draws a comparison between PBC and Non-adaptive modulation and supporting techniques (such as Matched Filtering (MF), Wavelet Modulation (WM)) that is the most associated method in the literature. A performance measure function refers to minimization of the area under the BER vs. E_b/N_0 graph. PBCS is implemented in semi-hardware platform to prove the applicability of the system. These results show that 1-bit encoded PBC demonstrates higher overall performance than the BPSK combined MF and WM in AWGN channel.

Another part of this study is regarding to optimize the spectral band utilization through a dynamic structure. It can respond to the changing user expectation in line with the Adaptive Modulation and Coding (AMC) principles. The user can construct distinct glossaries (waveforms) that can work with different data rates and under different noise levels. This capability provides the user with the flexibility to operate the communication line according to the specific preferred values. PBCS gives the advantage to select the most suitable point among different possible combinations in the SNR vs. LSE space. For this adaptive approach, PBC is compared with Quadrature Amplitude Modulation (QAM) to evaluate the adaptive performance of it. The similar tests are performed for Orthogonal Frequency Division Multiplexing (OFDM) and Pulse Shape Modulation (PSM) based modulation methods in simulation platform for evaluation the applicability of the proposed method in CR applications. According to simulation test results, PBC has between 3 to 6 dB better performance in adaptive application and it has between 1.5 to 4 dB better performance against to OFDM and PSM in AWGN channel.

Since the results are gathered in simulation environment, the next task is to realize these results in the hardware environment. Practically, there are two significant problems in the study: dynamic antenna solution and high sampling rate depending on the choice of the central frequency. According to the literature, it is possible to obtain this antenna structure by using Direct RF sampling method. In terms of the sampling rate problem, the minimum sampling rate should be twice the central frequency according to the Nyquist Theorem. Since Direct RF Sampling method can map the utilized channel to baseband in frequency spectrum, bandwidth is going to be taken as reference instead of the central frequency. Therefore, it is sufficient that the sampling rate of the Analog / Digital Converter (ADC) be twice the utilized bandwidth. The evaluation of hardware implementation is located in this thesis and the applicability of the proposed modulation technique and the recognition layer is discussed. In brief, according to our preliminary works in FPGA platform, the system can be realized with some limited level glossaries in the existing technology.

Another potential future work can be described as Pattern Based Multiple Access (PBMA). The pre-work on semi-hardware platform was performed in this thesis for this purpose. In addition to this, the cognitive engine can be extended and the network will be controlled cognitively. In this case, the system can be called Cognitive Network with multiple access capability.

The approach in this thesis is a new perspective in modulation and coding area of communication. Therefore, all design considerations of communication lines such as Multipath Effect, Doppler Effect, channel estimation, multiple accessing and network scheduling etc. could be handled in the future. Since this research area is wide, these studies are defined as a future work, but the solution advices for them have already mentioned in this thesis with some simulation tests.

The publications derived from this work also cited in survey (He et al., 2010) about the position of artificial intelligence in cognitive radio applications. Moreover, the published papers including the studies in this thesis are cited in several publications and dissertations either Ph.D. or M.Sc. (Taj, 2011, Taj et al., 2011, Taj et al., 2010, Kaminski, 2012, Mateo et al., 2012, Benidris et al., 2012, Tsagkaris et al., 2012). Last but not least, this work is also cited by one of the book published by Wiley (Grace et al., 2012).

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