CQI: Estimation Over The Dowlink of OFDMA

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Abstract

In this work, we analyze the problem of CQI measurements in the context of OFDMA which is the most probable candidate for DL of 3GPP-LTE. OFDMA will be employing link adaptation (adaptive coding and modulation, HARQ etc.) and frequency scheduling to mitigate the effects of inter-cell interference. These operations of AMC (Adaptive Modulation and Coding) and frequency scheduling will be performed on the basis of some feedback provided by the user to the base station. So our objective is to find some metric we can use as CQI feedback, measurement or estimation of this metric and proper schemes of feedback, over UL avoiding degradation of UL throughput.

We establish the fact that effective SINR, which is geometric mean of the SINR over symbols of sub-frame, can work well as CQI. Now the problem comes out to be the estimation of SINR over OFDM symbols. So in this thesis, we analyze different SINR estimation algorithms and discuss their suitability for our particular case of DL OFDMA. In the end, we come up with the most suitable algorithm for estimating SINR. The performance of this algorithm is analyzed with DL OFDMA system.

CHAPTER 1

Introduction

1.1 Context

The common feature of the next generation wireless technologies will be the convergence of multimedia services such as speech, audio, video, image and data. This implies that a future mobile wireless user will have guaranteed high-speed data access, will be able to enjoy infinite multimedia applications without constraints of its location and mobility and will be able to connect to different networks in order to have access to various services.

Besides offering new services with higher capacity and data rates, the goal of 4G systems is also to integrate the existing technologies in a common platform and to support multiple classes of terminals. Hence, a new generic high performance technology is required for future 4G systems, starting with a new generic physical layer.

At the beginning of the 1990s, the first generation of analog mobile radio systems, was replaced by the current second generation (2G) digital systems, such as GSM, IS-95 and PDC. Basically, the 2G digital systems were designed and optimized to provide speech and low-rate data services. The evolution is still continuing today with the deployment of the third generation (3G) systems, such as UMTS and CDMA-2000. The objectives of 3G systems went far beyond 2G systems, especially with respect to the wide range of multimedia services, high and variable data rates, wide and scalable bandwidth, high QoS requirements, operability in different environments, and flexibility in radio resource management. While 3G systems are currently being deployed, there has been already a significant research activity towards what is frequently referred to as beyond 3G or 4G systems. The rapid growth of internet services and increasing interest in portable computing devices are likely to create a strong demand for new wireless multimedia services requiring higher data rates than what is provided by 3G systems. Especially in the downlink, higher data rates are needed to accommodate emerging and new multimedia services such as video streaming, but also large data files downloading.

The most promising technology for future broadband downlink access is Orthogonal Frequency Division Multiple Access (OFDMA), which combines Orthogonal Frequency Division Multiplexing (OFDM) modulation and Frequency Division Multiple Access (FDMA) technique. The main advantages of OFDM multi-carrier modulation are its robustness in frequency selective channels and its low-complexity receivers thanks to equalization in the frequency domain. The advantages of OFDM multi-carrier modulation on the one hand and the zero intra-cell interference benefit of frequency division multiple access technique on the other hand make OFDMA a promising technology for broadband 4G systems in the downlink. In the recent years, there has been a very strong focus on this technology, including the realization of prototypes and field trials.

In this report, we focus on the issue of CQI measurements for DL of OFDMA which is required as it provides the basis for frequency scheduling and link adaptation done by the base station, for making possible high data rates and interference free transmission to users.

1.2 Contents of the Thesis Report

This section describes the contents of the next chapters so that reader has no problem in understanding the problem and its solution proposed later.

Second chapter is ’OFDMA Basics’. Here, first of all we give brief introduction of multi-path channel where we describe the propagation mechanisms of radio propagation and their effects on signal transmission. Then we outline the basics of OFDM modulation scheme describing its principle and then simple OFDM model which is frequently used in literature. We discuss also pros and cons brought to us by this multi-carrier modulation of OFDM. This chapter turns towards commonly used multiple access techniques and briefly describes FDMA, TDMA, and CDMA.

Third chapter named as 'Link Adaptation', gives us details for different forms of link adaptation mainly scheduling, adaptive modulation and coding and hybrid automatic repeat request. These have been used in HSDPA and will be used in next generation wireless networks because in literature they have been proved very promising for optimizing system throughput and spectral efficiency. Therefore this chapter gives us understanding for all of these techniques, what are basic gains in each technique and what are their requirements? This shows the reader that all link adaptation techniques depend upon the knowledge of channel quality of user. Thus need to have feedback from the receiver becomes very clear at this point and then we discuss different feedback techniques from receiver to transmitter. So in the end different feedback techniques have been presented alongwith their analysis and comparison.

Fourth chapter of 'CQI Measurements' has been devoted for discussion how to calculate an effective CQI. In the beginning, it offers us how to map CQI over some other metric which we are able to measure or estimate at user's side. Then it shows that geometric mean SINR of data symbols in a sub-frame can act as such a metric. In the end, this chapter presents our strategy for SINR estimation over different chunk and provides numerical results obtained after simulations.

CHAPTER 2

Basics of OFDMA

This chapter gives a brief overview of OFDMA for basic understanding and it constitutes a background for further discussions in this thesis report. OFDMA is a combination of Orthogonal Frequency Division Multiplexing (OFDM) and Frequency Division Multiple Access (FDMA). OFDM helps to establish high data rates even through multi-path fading channels. To understand fully OFDMA, it would be better first to have a brief introduction to mobile radio channel.

2.1 Mobile Radio Channel

In mobile radio communications, the path taken by signal between the transmitter and receiver antenna is referred to as the mobile radio channel.

2.1.1 Basic Mechanisms of Radio Propagation

The three basic propagation mechanisms are attributed as reflection, diffraction and scattering. All three of these phenomena cause radio signal distortions and give rise to signal fades and propagation losses.

Reflection occurs when a propagating electromagnetic wave impinges upon an object which has very large dimensions when compared to the wavelength of the propagating wave. Reflections occur from the surface of the earth and from buildings and walls.

Diffraction occurs when the radio path between the transmitter and receiver is obstructed by a surface that has sharp irregularities (edges). The secondary waves resulting from the obstructing surface are present throughout the space and even behind the obstacle, giving rise to a bending of waves around the obstacle, even when a line-of-sight path does not exist between transmitter and receiver. At high frequencies diffraction, like reflection, depends on the geometry of the object, as well as the amplitude, phase and polarization of the incident wave at the point of diffraction.

Scattering occurs when the medium through which the wave travels consists of objects with dimensions that are small compared to the wavelength, and where the number of obstacles per unit volume is large. Scattered waves are produced by rough surfaces, small objects or by other irregularities in the channel. In practice, foliage, street signs and lamp posts induce scattering in a mobile radio communications system.

2.1.2 Effects of Propagation Mechanisms

The three basic propagation mechanisms namely reflection, diffraction and scattering as we have explained above affect on the signal as it passes through the channel. These three mutually independent, multiplicative propagation phenomena can usually be distinguished as large-scale path loss, shadowing and multi-path fading.

Path Loss is the attenuation undergone by an electromagnetic wave in transit from a transmitter to a receiver in a telecommunication system. In simple words, it governs the deterministic average attenuation power depending only upon the distance between two communicating entities. It is considered as large scale fading because it does not change rapidly.

Shadowing is the result of movement of transmitter, receiver or any channel component (obstacles). Shadowing is not deterministic but a statistical parameter. Shadowing follows a log-normal distribution about the values governed by path loss. Although shadowing depends heavily upon the channel conditions and density of obstacles in the channel, it is also normally considered a large scale fading component alongside path loss.

Multi-path Fading is the result of multiple propagation paths which are created by reflection, diffraction and scattering. When channel has multiple paths, it is said to have memory. Each of the paths created due to these mechanisms may have its characteristic power, delay and phase. So receiver will be receiving a large number of replicas of initially transmitted signal at each instant of time. The summation of these signals at receiver may cause constructive or destructive interferences depending upon the delays and phases of multiple signals. Due to its fast characteristic nature, multi-path fading is called small scale fading.

2.1.3 Multi-path Channel Model

Multi-path channel model is a mathematical model which accounts for all the effects of multi-path radio propagation.

When a signal is transmitted in a channel, the channel acts like a filter. So the received signal becomes the convolution of transmitted signal and the channel impulse response. Thus it is represented as:

$$r(t) = h(t,\tau) \otimes s(t) = \int_{t=0}^{\infty} h(t,\tau)s(t-\tau)d\tau \qquad (2.1)$$

$$= \sum_{k=1}^{K} g_k(t) s(t-\tau_k) \qquad (2.2)$$

In above equation, h (t,τ) represents the channel impulse response. And $g_k(t)$ represents the effect of attenuation and phase of the k^{th} path changing with time t, and K is the total number of paths in the channel. With the help of equation 2.2, the channel impulse response h(t, τ) can be shown to be:

$$= \sum_{k=1}^{K} g_k(t) s(t-\tau_k) \qquad (2.3)$$

This equation shows the impulse response of a time variant channel. If we are dealing with a time invariant channel, where attenuation and phase associated with each of the K paths remain constant, the above equation gets modified to the simple form as given below:

$$h(\tau) = \sum_{k=1}^{K} g_k \delta(\tau - \tau_k) \tag{2.4}$$

2.2 Orthogonal Frequency Division Multiplexing (OFDM)

Orthogonal Frequency Division Multiplexing (OFDM) is an efficient multi-carrier modulation that is robust to multi-path radio channel impairments. It has been widely deployed in various high data rate multimedia communication standards like European standards for Digital Audio Broadcasting (DAB) and Terrestrial Digital Video Broadcasting (DVB-T), and the Wireless Local Area Network (WLL) standards such as IEEE802.16a. Now-a-days it is widely accepted that OFDM is the most promising candidate in future high data-rate broadband wireless communication systems.

2.2.1 Principle of OFDM

OFDM multi-carrier modulation aims at transmitting data at high rates while avoiding the Inter-Symbol Interference (ISI) caused mainly by multi-path propagation. To achieve this condition of zero ISI, OFDM modulation divides the initial serial high data rate stream of short symbol duration T_d into N_c parallel sub-streams of lower rate and thus of longer symbol duration T_s. The longer symbol time T_s compared to the maximum delay τ_{max} of the multi-path channel significantly reduces the inter-symbol interference (ISI). Each parallel sub-stream is modulated on one specific sub-carrier and adjacent sub-carriers are chosen with minimum frequency spacing $\Delta f = 1/T_s$ that is necessary to achieve orthogonality between the corresponding signals, presuming a rectangular pulse shape. The N_c sub-carrier frequencies are given by:

$$f_n = n\Delta f = nT_s \qquad ; n = 0 . . . N_c - 1 \tag{2.5}$$

Because we want to cancel out the ISI and preserve orthogonality between the sub-carriers, a guard interval of duration $T_g \geq \tau_{max}$ is inserted in the beginning of each OFDM symbol. So overall OFDM symbol duration exceeds to;

$$T'_s = T_s + T_g. \quad (2.6)$$

2.2.2 The Importance of Orthogonality

The main concept of OFDM is the orthogonality of sub-carriers. Since the carriers are all sine/cosine waves, we know that the area under one period of a sine or a cosine is zero, as shown in fig-2.1

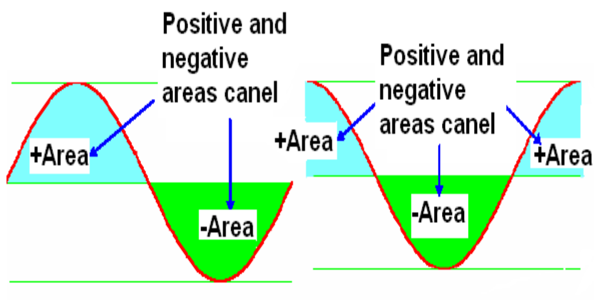

Fig-2.1 The area under a sine and a cosine wave over one period is always zero.

If we take a sine wave of frequency m and multiply it by a sinusoid (sine or cosine) of a frequency n, where both m and n are integers, the integral or the area under this product is given by:

$$f(t) = \sin m\omega t \;*\; \sin n\omega t \quad (2.7)$$

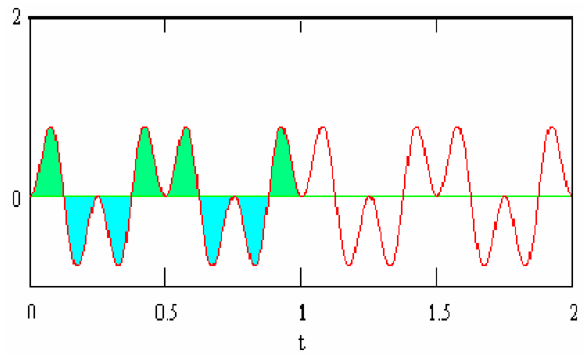

Fig-2.2 shows the signal f (t). The area under a sine wave multiplied by its own harmonic is always zero.

By the simple trigonometric relationship, this is equal to a sum of two sinusoids of frequencies (n-m) and (n+m), so f (t) becomes

$$h(\tau) = \sum_{k=1}^{K} g_k \delta(\tau - \tau_k) \tag{2.8}$$

These two components are each a sinusoid, so the integral is equal to zero over one period.

$$\int f(t)\ dt = \int_0^{2\pi} \frac{1}{2}\cos(m-n)\ \omega t dt - \int_0^{2\pi} \frac{1}{2}\cos(m+n)\ \omega t dt \tag{2.9}$$

$$= 0 - 0$$

We conclude that when we multiply a sinusoid of a frequency n by a sinusoid of frequency m/n, the area under the product is zero. In general for all integers *n* and *m*, sin *mx*, sin *nx,* cos *nx and* cos *mx* are all orthogonal to each other. These frequencies are called harmonics. This is a key idea in the concept of OFDM. The orthogonality allows simultaneous transmission on a lot of sub-carriers in a tight frequency space without interference from each other.

2.2.3 OFDM a Special Case of FDM

OFDM is a special case of Frequency Division Multiplexing (FDM). As an analogy a FDM channel is like water flow out of a faucet, in contrast OFDM signal is like a shower. In a faucet all of the water comes out as one big stream, putting thumb over the faucet hole would disrupt the complete water flow. But in case of shower putting thumb on it would not disrupt the entire flow. The response of OFDM to interference in comparison with FDM is also similar.

Fig-2.3

a) A regular FDM signal with all data on a single big data stream. b) An OFDM signal with multiple parallel bit streams carrying same amount of data.

In FDM if we have a bandwidth that goes from a to b, it can be divided into equal spaces, in fig. 2.3 it is divided into four equal channel spaces. Fig.2.4 shows frequency modulated carriers in frequency domain. The frequencies a and b could have any value, moreover the carrier frequencies do not have any specific relationship with each other.

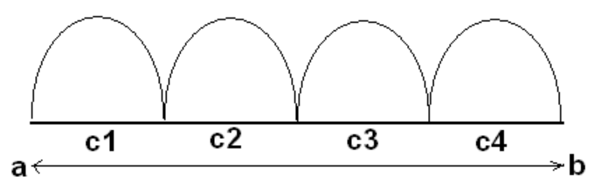

Fig-2.4

FDM carriers placed next to each other.

But if frequency *c1* and *cn* were such that for any integer n the following relationship holds:

$$Cn = n \times C1 \qquad (2.10)$$

So that

$$C2 = 2\ C1$$
$$C3 = 3\ C1$$
$$C4 = 4\ C1$$

Then all of these frequencies are harmonics of c1. In this case, since these carriers are orthogonal to each other, when added together they do not interfere with each other. In FDM the carrier frequencies do not follow the above relationship so they get interference from neighboring carriers. To prevent adjacent channel interference the signals are moved farther apart.

2.2.4 Simple OFDM Model

If we assume that guard interval T_g is greater than the maximum delay of multi-path channel τ_{max}, then guard interval will completely absorb the ISI as all symbols from different paths will arrive before the end of guard interval T_g. If we also suppose that channel response h (t, τ) is time invariant during this ISI free time interval $[iT_s' + T_{g,} (i+1)T_s']$ then received symbol in this time duration can be expressed as

$$r_n[i] = h_n[i]s_n[i] + v_n[i] \tag{2.11}$$

where $h_n[i]$ stands for channel frequency response at the n^{th} sub-carrier frequency fn during the time interval $[iT_s' + T_{g,} (i+1)T_s']$ and $v_n[i]$ represents the AWGN contribution.

This equation represents the simplistic OFDM model which represents OFDM communication as discrete transmission of symbols in time and frequency with N_c parallel Gaussian channels, each having its particular frequency response $h_n[i]$.

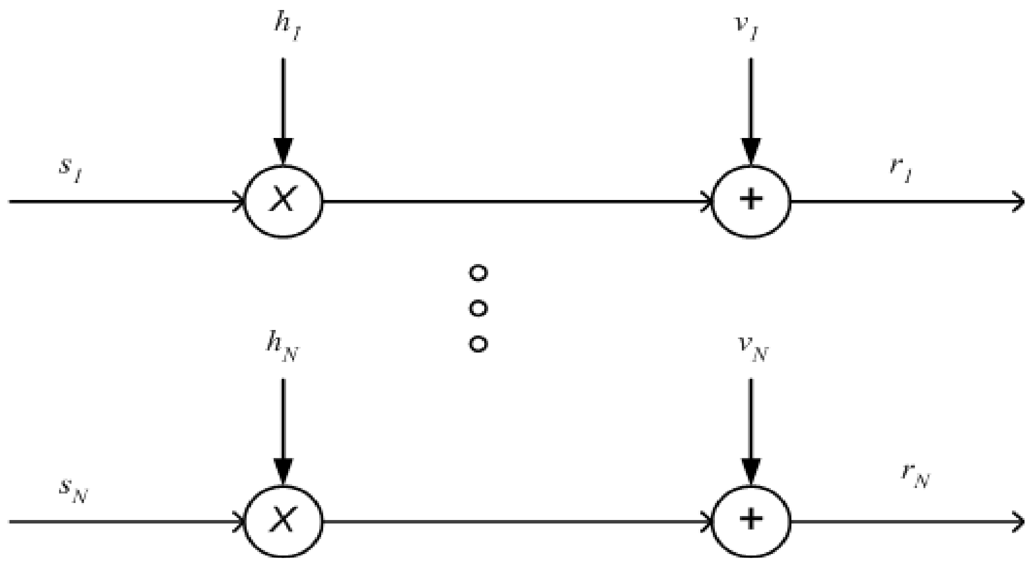

Figure 2.5 represents the simple OFDM model graphically.

2.2.5 Advantages and Challenges offered by OFDM

Advantages

OFDM has offered a lot of advantages as compared to single-carrier transmission schemes.

- OFDM is a highly bandwidth efficient scheme because different sub-carriers are orthogonal but they are overlapping.

- OFDM is an efficient way to deal with multi-path; for a given delay spread, the implementation complexity is significantly lower than that of a single carrier system with an equalizer.

- Flexible and can be made adaptive; different modulation schemes for sub-carriers, adaptable bandwidth/data rates possible.

- Has excellent ICI performance because of addition of cyclic prefix.

- In OFDM, equalization is performed in frequency domain which becomes very easy as compared to the time domain equalization.
- Very good at mitigating the effects of delay spread.

  - Due to the use of many sub-carriers, the symbol duration on the sub-carriers is increased, relative to delay spread.

  - ISI is avoided through the use of guard interval.

Challenges

There are some difficulties which should be overcome to use OFDM.

- The OFDM signal suffers from a very high peak to average power ratio (PAPR) therefore it requires transmitter RF power amplifiers to be sufficiently linear in the range of high input power.

- Sensitive to carrier frequency offset, needs frequency offset correction in the receiver.

- The use of guard interval to mitigate ISI affects the bandwidth efficiency.

2.3 Multiple Access Schemes

Cellular systems divide a geographic region into cells where a mobile user in each cell communicates with a base station. There are several different ways to allow multiple users the access to the channel which are called techniques of multiple access. The well known multiple access techniques are Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), Code Division Multiple Access (CDMA) and random access techniques.

2.3.1 Frequency Division Multiple Access (FDMA)

In FDMA, the whole bandwidth of the system is divided into smaller sub-bands and then base station allots these sub-bands to each user on demand. During the period, when one user is using one particular sub-band, no other user in the same cell uses this sub-band. In FDMA after the assignment of sub-bands for transmission and reception, the base station and mobile transmit simultaneously and continuously.

2.3.2 Time Division Multiple Access (TDMA)

TDMA systems divide the transmission time interval into time slots and in each slot only one user is allowed to either transmit or receive data. Thus when one user occupies time slot, it uses the whole bandwidth associated with this time slot. TDMA systems transmit data in a buffer-and-burst manner thus the transmission for any user is non-continuous. This makes TDMA a suitable scheme to use with digital data and digital modulation. In TDMA systems, synchronization overheads are sufficiently high because of bursty nature of transmission and in addition guard intervals are also required to separate bursts of two users.

2.3.3 Code Division Multiple Access (CDMA)

The multiplexing technique is based on the user-specific code. A pseudo-noise (PN) sequence converts a narrowband signal to a wideband noise-like signal before transmission. CDMA also provides immunity to multi-path interference and robust multiple access capability. CDMA systems become highly bandwidth efficient when used by multiple users but they are inefficient for a single user. There are two main types of CDMA techniques, frequency hopped multiple access and direct sequence spread spectrum.

2.4 OFDMA

OFDMA uses the combination of OFDM as modulation technique and FDMA as multiple access technique. It is also found in the literature as multi-carrier FDMA or multi-user OFDM. So in OFDMA multiple access is realized by providing each user with a fraction of the available number of sub-carriers. But it avoids the relatively large guard bands to separate different users which are necessary in FDMA. OFDMA uses dynamic resource allocation to give better performance, and to combat the multi-path channel effects and inter-cell interference.

2.5 OFDMA Downlink System

This section gives a brief description of OFDMA system in the DL direction. This is the system of interest which corresponds to a basic OFDMA system. So we describe the transmitter and receiver structure for this system.

2.5.1 Transmitter Structure

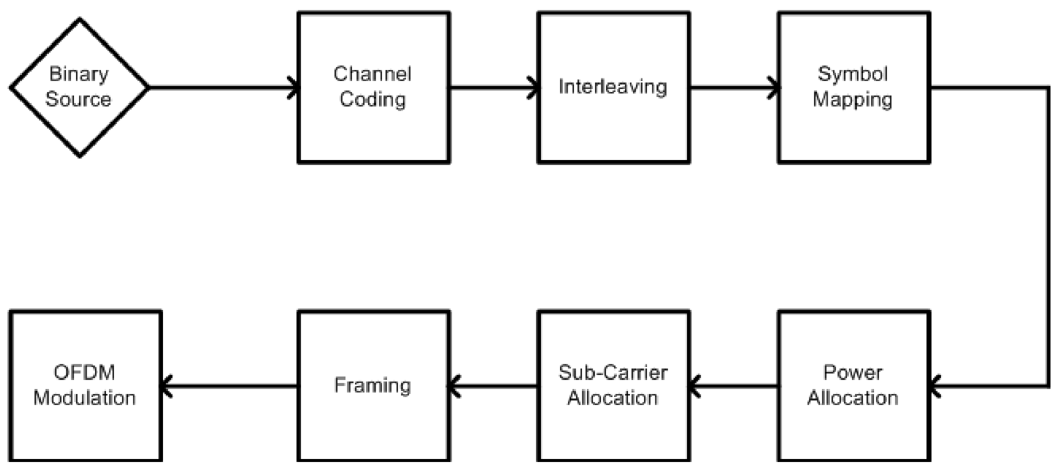

Figure 2.6 Transmitter Block Diagram of OFDMA

Figure 2.6 describes the block diagram of the base-band OFDMA base station (BS) transmitter. We suppose that number of active users served by BS is *K*. Information data streams for each user are individually coded and then interleaving is performed. Resulting interleaved coded bits are mapped to data symbols according to the modulation chosen. Then BS allocates different powers to data of each user. After power allocation, BS assigns sub-carriers to data of different users. The next step is to perform framing and adding pilot symbols at specified locations of sub-frame. Now this is the block of OFDM modulator which will perform the IFFT operation and also insert the guard interval. In the end, the complex envelope signal is RF modulated and then sent through the multi-path channels to users.

2.5.1.1 Channel Coding

When data is transmitted over a noisy channel, errors will occur. The role of channel coding is to minimize the number of errors in the transmitted data when it is decoded at the receiver. It is quite apparent that to perform this error detection and error correction task, channel coding needs to introduce redundancy in the data in a controlled manner so that later at the receiver side, exploitation of this added redundancy will help minimize the number of errors. So, coded sequences always have length greater than the original information sequences to perform error correction. The encoding process generally takes a block of k information bits and maps it into a block of n coded bits, called a codeword. The amount of redundancy introduced by the channel encoder is measured by the ratio $R_c = kn$, which is referred to as the code rate. Thus if the size of information stream is B_{info} and R_c is the code rate, then size of coded bit stream will be $B_{code} = B_{info}/R_c$

2.5.1.2 Bit Interleaving

In multi-path propagation environment, where fading often causes the signal to fall below the noise level over a large number of consecutive bits, bit errors normally occur in bursts. Most of the channel codes which are available to us behave with sufficient efficiency when the errors in the data are statistically independent. And they can not handle the situations properly when errors occur in bursts, which is the case in actual multi-path channel. To cope with bursty channel errors, interleaving is performed. At the transmitter side, interleaver performs an arrangement of all coded bits. When this data passes through the channel, errors of bursty nature are introduced in it. But at the receiver side, de-interleaver performs the reverse operation as that performed by interleaver. So it arranges the bits in their original positions. But this arrangement causes the bursty errors in the interleaved bits to spread randomly in the coded bits and breaks their bursty nature, making the code sufficiently efficient.

2.5.1.3 Symbol Mapping

The transformation of coded bit stream to complex valued data symbols is called Mapping. All possible complex valued data symbols form an alphabet of finite size M. A constellation diagram is most often used for illustrating the operation of mapping, as it shows the combinations of bits and their corresponding complex symbols along with their energy values. X-axis and Y-axis in the constellation diagram represent the real and imaginary axes for depicting complex symbols coordinates. There are different kinds of alphabets used for symbol mapping. The most common are Phase Shift Keying (PSK), Amplitude Shift Keying (ASK) and Quadrature Amplitude Modulation (QAM).

(i) Phase-shift keying (PSK)

Phase-shift keying (PSK) is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave).Any digital modulation scheme uses a finite number of distinct signals to represent digital data. In the case of PSK, a finite number of phases are used. Each of these phases is assigned a unique pattern of binary bits. Usually, each phase encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular phase. The demodulator, which is designed specifically for the symbol-set used by the modulator, determines the phase of the received signal and maps it back to the symbol it represents, thus recovering the original data.

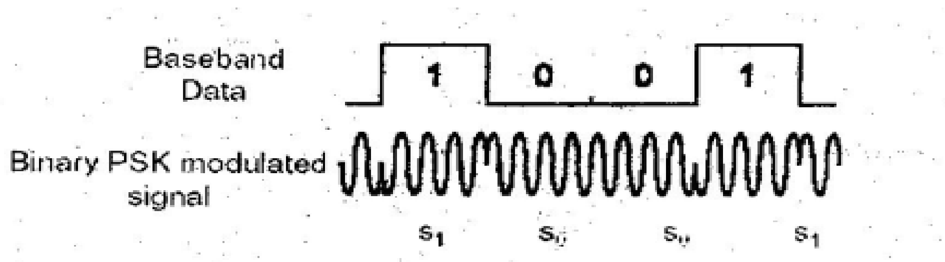

Figure 2.7: Phase Shift Keying

This requires the receiver to be able to compare the phase of the received signal to a reference signal — such a system is termed coherent. Phase-shift keying (PSK) is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave).

Any digital modulation scheme uses a finite number of distinct signals to represent digital data. PSK uses a finite number of phases, each assigned a unique pattern of binary bits. Usually, each phase encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular phase.

A convenient way to represent PSK schemes is on a constellation diagram. This shows the points in the Arg and plane where, in this context, the real and imaginary axes are termed the in-phase and quadrature axes respectively due to their 90° separation. Such a representation on perpendicular axes lends itself to straightforward implementation. The amplitude of each point along the in-phase axis is used to modulate a cosine (or sine) wave and the amplitude along the quadrature axis to modulate a sine (or cosine) wave.

In PSK, the constellation points chosen are usually positioned with uniform angular spacing around a circle. This gives maximum phase-separation between adjacent points and thus the best immunity to corruption. They are positioned on a circle so that they can all be transmitted with the same energy. In this way, the moduli of the complex numbers they represent will be the same and thus so will the amplitudes needed for the cosine and sine waves. Two common examples are "binary phase-shift keying" (BPSK) which uses two phases, and "quadrature phase-shift keying" (QPSK) which uses four phases, although any number of phases may be used. Since the data to be conveyed are usually binary, the PSK scheme is usually designed with the number of **Binary phase-shift keying (BPSK)** *(a) <u>Binary phase-shift keying (BPSK)</u>*

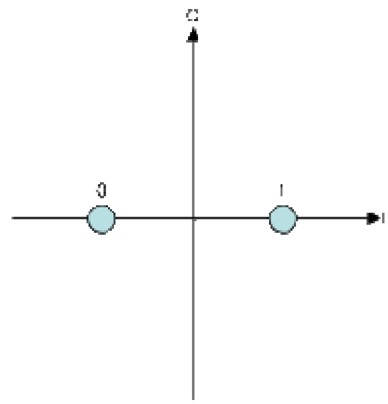

Figure 2.8: Constellation diagram for BPSK.

BPSK is the simplest form of PSK. It uses two phases which are separated by 180° and so can also be termed 2-PSK. It does not particularly matter exactly where the constellation points are positioned, and in this figure they are shown on the real axis, at 0° and 180°. This modulation is the most robust of all the PSKs since it takes serious distortion to make the demodulator reach an incorrect decision. It is, however, only able to modulate at 1 bit/symbol (as seen in the figure) and so is unsuitable for high data-rate applications when bandwidth is limited.

(b)<u>Quadrature phase-shift keying (QPSK)</u>

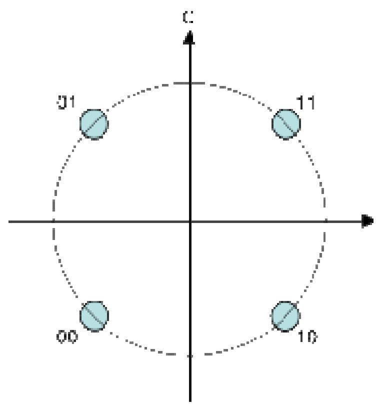

Figure 2.9: Constellation diagram for QPSK with Gray coding.

Each adjacent symbol only differs by one bit. Sometimes known as quaternary or quadriphase PSK or 4-PSK, QPSK uses four points on the constellation diagram, equi-spaced around a circle. With four phases, QPSK can encode two bits per symbol, shown in the diagram with Gray coding to minimize the BER — twice the rate of BPSK. Analysis shows that this may be used either to double the data rate compared to a BPSK system while maintaining the bandwidth of the signal or to maintain the data-rate of BPSK but halve the bandwidth needed.

Although QPSK can be viewed as a quaternary modulation, it is easier to see it as two independently modulated quadrature carriers. With this interpretation, the even (or odd) bits are used to modulate the in-phase component of the carrier, while the odd (or even) bits are used to modulate the quadrature-phase component of the carrier. BPSK is used on both carriers and they can be independently demodulated.

The implementation of QPSK is more general than that of BPSK and also indicates the implementation of higher-order PSK. The binary data stream is split into the in-phase and quadrature-phase components. These are then separately modulated onto two orthogonal basis functions. In this implementation, two sinusoids are used. Afterwards, the two signals are superimposed, and the resulting signal is the QPSK signal.

(ii) Quadrature amplitude modulation:

As with many digital modulation schemes, the constellation diagram is a useful representation. In QAM, the constellation points are usually arranged in a square grid with equal vertical and horizontal spacing, although other configurations are possible (e.g. Cross-QAM). Since in digital telecommunications the data is usually binary, the number of points in the grid is usually a power of 2 (2, 4, 8 …). Since QAM is usually square, some of these are rare—the most common forms are 16-QAM, 64-QAM, 128-QAM and 256-QAM. By moving to a higher-order constellation, it is possible to transmit more bits per symbol. However, if the mean energy of the constellation is to remain the same (by way of making a fair comparison), the points must be closer together and are thus more susceptible to noise and other corruption; this results in a higher bit error rate and so higher-order QAM can deliver more data less reliably than lower-order QAM, for constant mean constellation energy.

If data-rates beyond those offered by 8-PSK are required, it is more usual to move to QAM since it achieves a greater distance between adjacent points in the I-Q plane by distributing the points more evenly. The complicating factor is that the points are no longer all the same amplitude and so the demodulator must now correctly detect both phase and amplitude, rather than just phase.

64-QAM and 256-QAM are often used in digital cable television and cable modem applications. In the US, 64-QAM and 256-QAM are the mandated modulation schemes for digital cable (see QAM tuner) as standardized by the SCTE in the standard ANSI/SCTE 07 2000. Note that many marketing people will refer to these as QAM-64 and QAM-256. In the UK, 16-QAM and 64-QAM are currently used for digital terrestrial television.

(a)<u>Rectangular QAM</u>

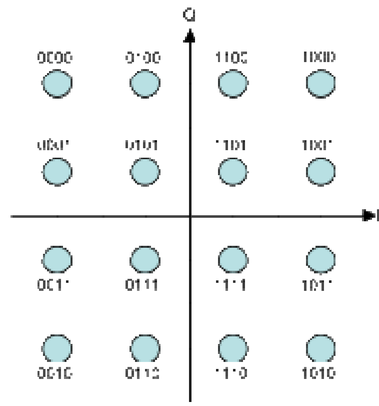

Figure 2.10: Constellation diagram for rectangular 16-QAM.

Rectangular QAM constellations are, in general, sub-optimal in the sense that they do not maximally space the constellation points for a given energy. However, they have the considerable advantage that they may be easily transmitted as two pulse amplitude modulation (PAM) signals on quadrature carriers, and can be easily demodulated. The non-square constellations, dealt with below, achieve marginally better bit-error rate (BER) but are harder to modulate and demodulate.

The first rectangular QAM constellation usually encountered is 16-QAM, the constellation diagram for which is shown here. A Gray coded bit-assignment is also given. The reason that 16-QAM is usually the first is that a brief consideration reveals that 2-QAM and 4-QAM are in fact binary phase-shift keying (BPSK) and quadrature phase-shift keying (QPSK), respectively. Also, the error-rate performance of 8-QAM is close to that of 16.

2.5.1.4 Power Allocation

According to the channel conditions, base station assigns suitable power levels to each user so that its demands can be satisfied. It also depends upon the distance of user from the base station.

2.5.1.5 Sub-Carrier Allocation

After power allocation, data of different users are multiplexed. The system concerned to us in this report is OFDMA which uses frequency division multiplexing so the symbols of different users need not be added as it is done with the chips of CDMA. So there is no overlapping of data symbols from different users. Sub-carrier allocation is mapping of data symbols of different users to particular frequency and time positions in OFDMA sub-frame. There are principally two strategies for arranging sub-carriers for different users [1]. One is Localized Frequency Mapping. In this scheme each user is assigned a set of sub-carriers which are adjacent in frequency to each other. Contrary to this, other scheme named Distributed Frequency Mapping, assigns each user sub-carriers which are located sufficiently away from each other in frequency. This method gives us again because of frequency diversity, as different sub-carriers which are farther in frequency fade independently of each other. Other schemes are also possible applying this strategy in time domain which do mapping in adjacent OFDM symbols or different OFDM symbols.

2.5.1.6 Framing

In Framing block different kinds of sub-carriers e.g. pilot sub-carriers and null sub-carriers are adjusted at their proper positions in OFDMA sub-frame. They make possible for users data acquisition, channel estimation and serve as guard carriers. Null sub-carriers are required in the system to act as guard bands to avoid any kind of interference to systems using neighbouring frequencies. The important task of framing is to put pilot symbols in their proper positions. Pilot symbols help the receivers in estimating the channel and synchronizing themselves with the data transmitted. Different patterns having special features can be used as pilot sequences.

2.5.1.7 OFDM Modulator

This block mainly performs operations of IFFT or IDFT followed by the addition of guard time interval.

2.5.2 Receiver Structure

Figure 2.11 depicts a block diagram of the base band OFDMA mobile station receiver.

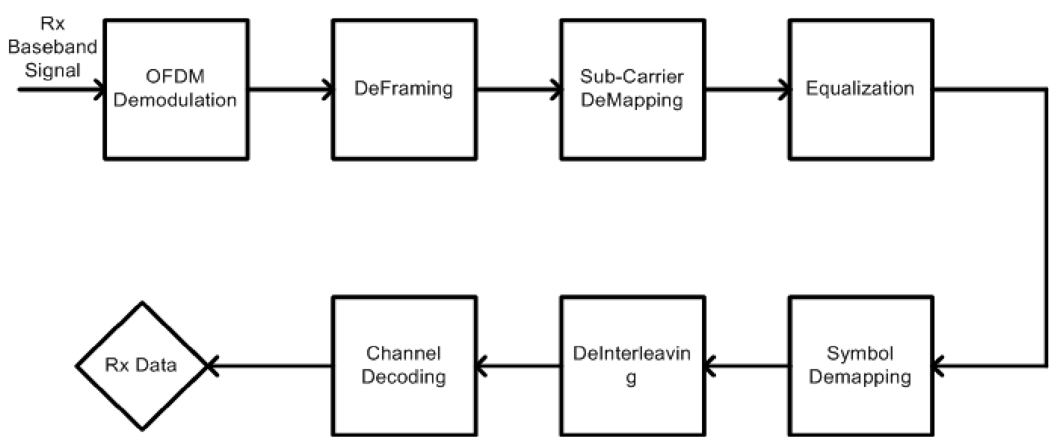

Figure 2.11: Receiver Block Diagram of OFDMA

At the receiver side, operations go in the reverse order as they are performed at the transmitter.

2.5.2.1 OFDM Demodulation

At the receiver, the received signal is first sampled and OFDM demodulation is performed by removing the guard time interval and FFT/DFT operation. And this is due to the presence of guard interval that we avoid all ISI.

2.5.2.2 De-framing

Receiver performs de-framing and gets pilot symbols at their proper positions. This helps user channel estimation which later serves in equalization. Pilots are also used for getting an estimate of channel quality and feedback may be sent.

2.5.2.3 Sub-Carrier De-mapping

In sub-carrier de-mapping a particular receiver separates its data from the data of all users sent by the transmitter.

2.5.2.4 Equalization

Detection and equalization are performed to get the stream of data symbols for this particular user. This is possible based on the channel estimates performed over pilots. There are various equalization strategies and one may be chosen based on the requirements.

2.5.2.5 Symbol De-mapping

Operation of De-mapping converts this data symbol stream to stream of interleaved soft bits.

2.5.2.6 De-interleaving

De-interleaving operation performs inverse of interleaving so that coded bits get their proper positions as they occupied when coding was done at the transmitter. Interleaving adds no redundancy to data but it helps to randomly spread the channel errors.

2.5.2.7 Decoding

Decoding converts the coded bits to groups of information bits sent by transmitter and this is final output for a receiver.

2.6 Conclusions

This chapter has provided us some basic information to understand our system and context of our work. First it gives us brief description of all phenomena of channel, how multi-path channel behaves and how we model this channel. Then it explains to us the most promising multi-carrier modulation technique, OFDM. Then we find description for multiple access techniques. Then it introduces us OFDMA in the DL direction which is the system of interest for our work. For OFDMA, it explains the basic structure of transmitter and receiver with their proper functioning. All of this information enables us to well understand the problem we will be dealing with, later in this thesis report.

CHAPTER 3

Link Adaptation

This chapter describes, what is link adaptation, what are its different forms, how it can be put into practice and what are the possible gains that we can realize by using different techniques for link adaptation.

3.1 Basics of Link Adaptation

Link adaptation is a term widely used in wireless communications to denote the matching of the modulation, coding and other signal and protocol parameters to the conditions on the radio link (e.g. the path loss, the interference due to signals coming from other transmitters, the sensitivity of the receiver, the available transmitter power margin, etc.). The process of link adaptation is a dynamic one and the signal and protocol parameters change as the radio link conditions change dynamically. Link adaptation has been successfully employed in HSDPA (High Speed Downlink Packet Access).

The main challenge for future wireless communication systems is to achieve the required performance for a broad range of applications with minimal utilization of resources. There may be bad channel conditions or bad interference conditions which result in a performance degradation that cannot be tolerated by future demanding multimedia applications. However, next to the propagation, environmental and interference conditions, also the data rate requirements of future applications are highly dynamic and variable. Hence, traditional systems, which have been designed and optimized for operation in one typical situation, which is only valid for a small percentage of the time, most of the time can not achieve the optimal performance with minimal resource utilization. Then on the other hand, some protocols have been devised which always meet some minimal data rate requirements. These are based on the worst channel and interference conditions. These are also not optimal because these worst conditions do not long for good. But they hold a small amount of time. So they also make wrong use of system resources. The best solution is to adapt channel and interference conditions in a manner to maximize the system efficiency.

3.1.1 Decision Parameters for Link Adaptation

There are a lot of parameters over which link adaptation decisions can be taken. But they depend effectively over the constraints of environment, user's demand and availability of system resources. In the Long Term Evolution of 3GPP, some parameters have been defined which should be taken into account for link adaptation in this case of DL OFDMA. These are as follow:

- QoS (Quality of Service) parameters namely BER (Bit Error Rate), FER (Frame Error Rate), min bit rate etc.

- Channel conditions measured by user.

- Interference conditions measured and indicated by user to base station.

- Capability of user equipment.

- System parameters such as bandwidth and interference level.

- Pending retransmissions for some user due to loss of earlier transmissions.

3.1.2 Different Link Adaptation Techniques

As our definition of Link Adaptation specifies that any change of parameter or modification in some technique to better adapt channel conditions, interference and demand is link adaptation. So there are lot of possibilities to implement link adaptation. But certainly there are only a small pile of strategies which are very commonly known and give us sufficient gain. These basic techniques are:

1. Scheduling of Resources.

2. Adaptive modulation and coding (AMC).

3. Hybrid Automatic Repeat reQuest (HARQ).

In the following sections, we provide details for all three of these techniques and then we discuss how to put these techniques into practice.

3.2 Scheduling

In a communication system, allocation (reservation) of resources (frequency or time etc) for different users depending upon environment, channel conditions, interference level and user's demands to maximize the throughput and system efficiency is called scheduling.

For a multi-user system where user channel conditions change over time, a scheduler can take advantage of channel variations by giving certain priority to the users with transitorily better channel conditions. Hence, the choice of the scheduling algorithm critically impacts the system performance. Several scheduling algorithms have been proposed in the literature to maximize the packet data throughput, subject to various fairness conditions.

In our concerned system of DL OFDMA, the scheduler of base station dynamically controls which time/frequency resources are allocated to a certain user, at a given time. Downlink control signalling informs the users what resources and respective transmission formats have been allocated. The scheduler can even instantaneously choose the best multiplexing strategy from the available methods e.g. frequency localized or frequency distributed transmission.

3.2.1 Round Robin Scheduler

Round Robin is one of the simplest scheduling algorithms, which has been used for processes in an operating system or for different users in a wireless network, which assigns resources to each probable candidate in equal portions and in order, handling all candidates without any priority. Round Robin scheduling is both simple and easy to implement, and starvation-free. Round Robin scheduling can be applied to any system where there is need to schedule the resource usage.

In context of wireless networks, this algorithm provides every user to transmit or receive on the shared channel at a regular interval. Round Robin may appear a fair algorithm in the sense that it provides equal resources to all users. But practically it is highly unfair algorithm because this algorithm does not take into account the changing reception conditions at the different receivers and thus it will schedule transmissions to/from subscribers half of the time when their reception conditions are worse than average. Similarly it is blind to user's demands and application requirements. So it casts a heavy blow to system throughput and spectral efficiency. The name of the algorithm comes from the round robin principle known from other fields, where every participant takes its equal share at its proper turn.

3.2.2 Proportional Fair Scheduler

Proportional fair scheduler is considered one of the best for resource allocation to multi-user systems. In contrast to round robin scheduler, the Proportionally Fair scheduler will schedule transmissions every time to/from all subscribers when the reception conditions are well above the average and will take into account application requirements. In wireless networks it works along with adaptive modulation and coding and HARQ. First of all scheduler receives some feedback from the desired user supposedly signal to interference plus noise ratio SINR and this feedback gives the base station scheduler a measure of channel and interference conditions. Upon the basis of this feedback, and user's demand and data requirement, scheduler selects a modulation and coding scheme for the user. After selecting an MCS for each user, the scheduler determines the user to be served in the future based on the instantaneous bit rate of each user. This scheduler gives approximately the same number of time slots to all the users, but assigns the transmission to each user when its channel condition is at its best.

3.3 Adaptive Modulation and Coding (AMC)

The core idea of AMC is to dynamically change the Modulation and Coding Scheme (MCS) in subsequent frames (packets) with the objective of adapting the overall spectral efficiency to meet the channel conditions and based on the parameters as we mentioned in section 3.1.1. The decision about selecting the appropriate MCS is performed at the transmitter side. But for this decision foremost importance is given to channel quality at the receiver. Thus it is the responsibility of receiver to estimate the channel quality (based upon the previous transmission) and then give it as feed back to the transmitter with proper latency. Many AMC techniques have been presented in the literature.

The implementation of AMC offers several challenges. First, AMC is sensitive to measurement errors and delays. In order to select the appropriate modulation, the scheduler must be aware of the channel quality. Errors in the channel estimate, will cause the scheduler to select the wrong data rate and either transmit at too high a power, wasting system capacity, or too low a power, raising the block error rate. Delay in reporting channel measurements also reduces the reliability of the channel quality estimate due to the constantly varying mobile channel. Furthermore changes in the interference add to the measurement errors.

Typically, in high speed downlink systems each transmission is at maximum power available and no power control is employed. Therefore, link adaptation, which continuously adjusts the modulation and coding scheme (MCS), provides an efficient way of maximizing the instantaneous usage of the wireless channel. For example, it enables the use of spectrally efficient higher order MCSs when channel conditions are favourable, while reverting to the MCSs that are more robust but with lower transmission rates when channel conditions degrade.

3.4 Hybrid Automatic Repeat reQuest (HARQ)

3.4.1 Simple ARQ Schemes

In the literature, there exist three most common simple ARQ schemes which are given as below.

3.4.1.1 Stop and Wait

This is the simplest of ARQ schemes. Here transmitter transmits one data packet or frame and then waits for its ACK or NAK to be sent from the receiver. In case of ACK, it goes for the transmission of the next packet to that receiver. If a NAK is received, it indicates the existence of an error in the previously transmitted packet and retransmission occurs. Normally if an ACK or NAK does not reach in a specific time, it is considered as reception of a NAK and transmitter retransmits.

3.4.1.2 Go Back N

In this scheme, transmitter transmits a packet and then does not wait for its response, rather it starts transmitting a second packet if it has one. In this case transmission remains continuous. But if NAK is received for some packet, that packet and all after that are retransmitted. This scheme performs better than simple stop and wait ARQ.

3.4.1.3 Selective Repeat

In selective repeat scheme, transmission is continuous like go back N scheme but if NAK is received for a particular packet, only that packet is retransmitted and all later packets are not retransmitted. Because there is a possibility that a packet got errors but later packets were received successfully by the receiver. This scheme requires transmitter and receiver both to maintain large buffers but it is most efficient in simple ARQ schemes.

3.4.2 HARQ

The performance of link adaptation largely depends upon the accuracy of channel quality measurements, which is difficult to maintain as mobility of user increases. When packet data is delay-tolerant, this quality makes it feasible to use retransmission schemes to recover erroneous packets. If we use simple FEC schemes, they require us to use powerful codes which have complex decoding and they don't use the system resources efficiently. So we will be degrading the spectral efficiency of the system. Simple ARQ schemes on the other hand, have their own disadvantages. When channel conditions are not promising, they degrade the system performance. The best solution for these kind of problems is to use the combinations of FEC schemes and ARQ schemes which are referred to as the Hybrid ARQ schemes. Hybrid Automatic Repeat reQuest (HARQ) techniques have been adopted by several wireless standards, e.g., HSDPA.

HARQ can compensate for link adaptation errors and provide a finer granularity of coding rate and improve throughput performance. Upon detecting a transmission failure, typically through cyclic redundancy check (CRC), the mobile user sends a request to the base station for retransmission. The packet decoder at the mobile may combine the soft information of original transmission with that of the subsequent retransmissions, and the combined signal will have higher probability of successful decoding.

3.5 Channel Quality Indicator and CQI Feedback

The quality of communications channel can have a significant impact on the performance of a wireless communications system. A communications system having channels with high channel quality can transfer data at a higher rate and with a lower latency than a communications system having channels with low channel quality. Given a pair of communications channels in a communications system, a communications channel with high channel quality will more likely support a higher maximum bandwidth as well as have a lower error rate than a communications channel with low channel quality. Channel quality should therefore be a prime factor that needs to be considered when performing any act of link adaptation. Upon receipt of the CQI (Channel Quality Indicator), the BS can use the CQI to schedule transmissions from the UE (and other UEs). The scheduling can involve the assignment of transmissions to certain channels as well as transmission order and time. In addition to scheduling transmissions based on the CQI, the BS can also make modulation and coding scheme (MCS) assignments for the channels.

3.5.1 What is CQI?

A channel quality indicator (CQI) can be a value (or values) representing a measure of channel quality for a given channel. Typically, a high value CQI is indicative of a channel with high quality and vice versa. A CQI for a channel can be computed by making use of some performance metrics, such as signal to noise ratio (SNR), signal to interference plus noise ratio (SINR), signal to noise plus distortion ratio (SNDR), and so forth of the channel. These values and others can be measured for a given channel and then used to compute a CQI for the channel. CQI may also be depending upon MCS used.

3.5.2 Interests for CQI Feedback

Channel quality indicator feedback from UE which may indicate the downlink channel quality and interference conditions can be used at the base station for at least the following purposes:

- Time/Frequency selective scheduling so that users can be scheduled for their best channel.

- Choice of MCS which corresponds to user's demand and is sufficiently rigorous to give QoS.

- Interference Management as if UE senses high interference conditions over some channels, it can be allocated some other group of carriers.

3.5.3 Computation of CQI

One commonly used technique to compute CQI is to determine a value of a metric for a channel and then use the value to compute the CQI. In other words we map some measurable value of channel as a CQI. There may be different measurable values we may use to obtain CQI depending upon the requirements and suitability. The CQI for the channel can then be used in a variety of operations involving the channel, such as scheduling transmissions on the channel.

3.5.3.1 CQI over Groups of Carriers

When we are dealing with multi-carrier communications, it is really difficult to have a CQI feedback value for each carrier. We can make groups of carriers depending upon the coherence bandwidth of the channel. So this group of carriers will be normally showing the same fading characteristics, which is governed by the coherence bandwidth of the channel. Thus state of the channel over this group of carriers will not be very different from each other. So, even one single value will be sufficient to represent the channel quality for this group of carriers. This will avoid the havoc over the uplink throughput by reducing the amount of feedback information.

Thus the use of a representative CQI value for a group of carriers can reduce CQI feedback requirements. This is in no way non-significant but it has another facet which is again promising. The channel conditions over a group of sub-carriers will not be much different depending upon the coherence bandwidth of the channel. So even we don't need to measure same value over all sub-carriers to get CQI for this group. We can just compute this value over one carrier and map it to CQI, knowing the fact that behaviour of group is similar. This will reduce the processing required at the receiver. On the other hand, this characteristic can be utilized another way. We can measure some value (say SINR) over all sub-carriers over the group and get one CQI value which better corresponds to this whole group. This way we will get a very refined value of CQI.

3.5.3.2 Group CQI and Interference

Another possible thing which should be taken care of that we may take some sub-carriers in a group and we are trying to calculate transmission quality over these carriers. And suppose that channel coherence bandwidth is sufficiently large so that fading characteristics are nearly the same over this group. Now the possibility may exist that out of these sub-carriers, some are in interference (may be inter-cell interference). So measurement over only one sub-carrier may lead the receiver to choose a CQI value which does not correspond to channel quality over this group of sub-carriers. So there must be some way to properly handle this kind of situations.

In the case of DL of OFDMA (the scenario we are dealing with in this project), we take a chunk composed of 25 sub-carriers. And whenever one BS selects a chunk for transmission, it will use all 25 sub-carriers. So, all sub-carriers in one chunk of this system will be in the same interference conditions. But this may pose a problem in the case if a BS is using localized transmission and its neighbouring BS is using distributed transmission technique. This will cause some carriers in one PRB (Physical Resource Block) to have interference and other carriers will be interference free.

3.5.3.3 CQI Requirement

The computation of a CQI for a channel (or channels) can begin with a transmission from the BS to the UE. The transmission may be a normal transmission of data from the BS to the UE, a transmission of control information, a special transmission intended solely to measure the quality of the channel(s), or so forth. With the reception of the transmission at the UE, the UE can obtain a measure of the channel quality. As an alternative to actually having the BS transmit to the UE, especially when the BS may not have anything to transmit to the UE, the UE may obtain a measure of the channel quality by measuring a designated channel, such as a pilot channel. The pilot channel is normally used by a UE to become synchronized with the multi-carrier communications system as well as obtain control information from the BS.

A BS can make use of the CQI values from the various UEs to schedule transmissions to the UEs. For example, the BS may select to place more transmissions on a carrier that has a high CQI value and fewer transmissions on a carrier that has a low CQI value. In addition to scheduling transmissions to the UEs, the BS can make MCS determinations for each of the carriers, using the CQI values. For example, for a carrier with a low CQI value, the BS may select to use an MCS that will provide high degree of tolerance to errors at the expense of data throughput to ensure that transmissions will be successfully received. While for a carrier with a high CQI, the BS may select to use an MCS that will minimize the overhead to maximize the data throughput. After the BS has performed scheduling and MCS determinations, the BS can commence transmissions to the UEs.

3.5.4 CQI Feedback Techniques

As long term evolution of 3GPP is considering OFDMA as the most suitable candidate for the DL of future wireless networks, there have been extensive research over the CQI feedback techniques in different committees of 3GPP. They have discussed a lot of CQI feedback techniques, the most notably of which we discuss below:

3.5.4.1 All Chunk CQI Feedback

According to this scheme, each UE should give CQI feedback for all of the chunks after each *x* sub frames. And each value should be the average CQI for all of the sub-carriers in one chunk. This technique is more suitable for localized transmission. But it also increases the overhead over UL control channel for sending this feedback to BS.

3.5.4.2 Top-M Chunk CQI Feedback

According to this scheme, each UE is not obliged to send feedback for all of the chunks in the system, rather it will send feedback only for M most suitable chunks. The reason is that except for very high load conditions, BS will be able to do scheduling of this UE over one of M chunks and assign MCS according to its quality in that chunk. This scheme performs much better than if we feedback for all chunks in terms of DL throughput and UL system overhead.

3.5.4.3 Average of Top-M Chunk CQI Feedback

In this scheme, each UE mentions top-M chunks according to reception conditions and along with this, indicates the average CQI value over these M chunks. So it reduces the feedback even from scheme where we give feedback of each of Top-M chunk.

3.5.4.4 Average CQI Feedback for Whole Bandwidth

This scheme demands from all users to give only one CQI value over the whole bandwidth of the system after each x sub frames where x may be chosen depending upon the channel coherence time and latency requirements. This scheme may work for an OFDMA system utilizing distributed transmission, because in distributed transmission, sub-carriers of each user are distributed over the whole ODMA bandwidth. But this technique is highly inefficient because only one value can not represent the quality of whole system bandwidth.

3.5.5 Comparison of Different CQI Feedback Techniques

There is a scheme based on the sending feedback after taking its DCT (Discrete Cosine Transform) but it is worse than Top-M average. Similarly there is Bitmap scheme where each chunk is represented by one bit and feedback for some chunks is sent and for others it is some value below the value sent.

3.6 Conclusions

This chapter deals with the concept of link adaptation which is not very old for wireless networks. Here in the beginning, the notion of link adaptation has been defined. Then later sections give a detailed picture of different forms of link adaptation, namely scheduling, AMC and HARQ. Each section gives stress on the quality of UE feedback which will become the basis for link adaptation. So the need to have an explicit feedback from the UE becomes very primordial. In the end of chapter, we explain whole about CQI feedback, its utilization, its implementation and feedback techniques. This chapter has provided us evidence that channel quality estimation is really very necessary and it will pay a lot in terms of system throughput and spectral efficiency. Thus our next chapter will deal with this idea of estimation of quality of channel to realize all of these gains we have discussed in this chapter.

CHAPTER 4

CQI Measurements

Chapter 3 of 'Link Adaptation' showed us what are the possible processes at the transmitter where knowledge of channel quality at the receiver may be used and what are the corresponding possible gains? Thus receiver should provide the transmitter with the feedback about the transmission quality (channel quality and interference conditions), so that the transmitter is able to take Link Adaptation decisions over this feedback. In this chapter we investigate what are the measures or metrics a receiver can have, and use them to map as CQI. Then it details how these measurements can be made efficiently.

4.1 Problem Statement

As is clear to us that user should provide some feedback to BS corresponding to the quality of transmission, so that BS transmitter can take adequate decisions for mechanisms of link adaptation namely, frequency scheduling, AMC and HARQ etc. Now we should find some information at the user side which reflects the quality of transmission. Frame Error Rate (FER) or Frame Error Probability (FEP) is the measure which may describe the quality of transmission. The throughput is also a good measure of quality of transmission but FER maps directly over throughput. Keeping in mind the fact that a user does not have access to data transmitted to it, it is unable to calculate ideal FER, but it can estimate FER. Estimation of FER is done by using some model of quality which has the general form:

$$FER = LUT\,(CQI) \qquad ; CQI = f(Q_1, Q_2, \ldots, Q_N) \tag{4.1}$$

Where LUT (Look-Up Table) is a table of correspondence which maps the values of CQI feedback to FER and it is specific to each MCS. Now CQI is the metric of quality which user should provide to the transmitter of BS. CQI is a function f of instantaneous values $\{Q_n\}$ over the duration of sub-frame or the interval over which system takes link adaptation decisions. Function *f* can also be dependent upon MCS in a soft way. The variables $\{Q_n\}$ are functions of propagation and interference conditions which affect this particular transmission. So we can write:

$$Q_n = g(h_n, \sigma^2_{N+1}) \tag{4.2}$$

where h_n and σ^2_{N+I} represent respectively power of signal (propagation effect) and variance of noise plus interference. These variables can be combined in a single metric of SINR (Signal to Interference and Noise Ratio). So Q_n becomes:

$$Q_n = SINR_n = \frac{|hn|^2}{\sigma^2_{N+1}} \tag{4.3}$$

As data symbols in a sub-frame are transmitted over different sub-carriers, so they may experience different propagation and interference conditions. Hence SINR values will differ. So we have to compress N values corresponding to N data symbols into one value and this one value should represent the quality of transmission for this set of data symbols. We call this single value as $SINR_{eff}$. The first solution that comes to mind is the arithmetic average of these N values, so:

$$SINR_{eff} = \frac{1}{N}\sum_{n=1}^{N} SINR_n \tag{4.4}$$

where $SINR_n$ denotes the instantaneous SINR for the n^{th} detected symbol in the frame. This solution typically overestimates the SINR because arithmetic average will favor the higher SINR values. The second simple solution is to calculate geometric mean of the N values, which is given by

$$SINR_{eff} = \left(\prod_{n=1}^{N} SINR_n\right)^{\frac{1}{N}} \tag{4.5}$$

The geometric average is, in fact average of logarithm of SINR values. This geometric mean solution gives results better as compared to the arithmetic mean.

The calculation of $SINR_{eff}$ using geometric mean gives sufficiently accurate results. They give very good results for QPSK and acceptable results for 16QAM.

And these are modulation schemes we will be working with, in our simulation environment, so we will use geometric mean as a compression function for instantaneous SINR values over the sub-frame.

We have seen that our original motivation of realization of gains associated with link adaptation required some CQI feedback. This CQI feedback, as we have decided, will be the geometric mean of instantaneous SINR over symbols of one sub-frame so the only problem left unsolved to us is the estimation of instantaneous SINR over the symbols of one sub-frame and this is the problem, we will concentrate over in the rest of this chapter.

4.2 Considerations for Estimation of SINR

Until this point, we have come to realize that a receiver needs to estimate instantaneous SINR, get its effective value and return it as CQI (Channel Quality Indicator) feedback to the transmitter. Thus system may realize the gains related with link adaptation, and particularly in our system of DL OFDMA, this will be used as a major technique to avoid inter-cell interference. Before we start tackling the problem of estimation of SINR, there are some small but tricky points which need to be clarified.

4.2.1 Concept of PRB and VRB

In DL of OFDMA users are assigned sub-carriers. This sub-carrier assignment can be Localized or Distributed as described in section 2.5.1. From this we get two terms PRB (Physical Resource Block) and VRB (Virtual Resource Block). PRB is a set of sub-carriers consecutive in frequency so they are always adjacent to each other. VRB is a set of sub-carriers which are allocated to one particular user. So the VRBs are sometimes Localized and sometimes Distributed for frequency diversity.

When sub-carrier mapping is Localized, VRB is mapped over PRB/PRBs in a localized manner, so it becomes a set of consecutive sub-carriers. When sub-carrier mapping is Distributed, VRB is mapped over multiple PRBs in a distributed manner so each VRB has some sub-carriers in multiple PRBs.

4.2.2 Need for Instantaneous SINR

This fact is much important about SINR that we don't need average SINR over a long interval of time. SINR which is fed back by the receiver to the transmitter is used in frequency scheduling, interference avoidance and AMC. Thus it binds the receiver to send instantaneous values of SINR to the transmitter as feedback so that the transmitter is able to allocate optimal resources to this particular receiver on the basis of instantaneous quality of transmission. In 3GPP-LTE, the time period of this feedback interval has not been specified yet but they have done a lot of experimentation of CQI feedback normally taking the feedback interval equal to one sub-frame (7 OFDM symbols/0.5 ms). It has been recommended that time granularity of CQI feedback should be adjustable in terms of sub-frame units and it may be set different per user basis or per group of user basis.

4.3 State of the Art SINR Estimation Algorithms

4.3.1 SINR Estimation Requirements

Before describing the state of the art algorithms for SINR estimation, it is better to clarify the definition of SINR. SINR just as abbreviation suggests is signal to interference and noise ratio.

$$SINR = \frac{Signal\ Power}{Noise\ Power + Interference\ Power} \tag{4.6}$$

$$SINR_n = \frac{|h_n|^2}{\sigma^2_{N+I}} \tag{4.7}$$

Equation 4.7 gives mathematical shape to the general equation just above and it makes following two implicit assumptions:

- It assumes that normalized symbols were transmitted by the transmitter so signal power is completely given by $|h_n|^2$.
- It assumes that noise power σ^2_{N+I} is same for the group of sub-carriers assigned to some particular user.

We are aware of the fact that channel coefficients which dictate signal power in this case are provided by channel estimator and they are already known.

4.3.2 Identical Sequence Method

The receiver receives a sequence of symbols. Later that same sequence is repeated from the transmitter. When user takes difference of these two received samples, assuming same noise and as data sequences are identical, UE gets difference of noise in two received samples. From this, receiver calculates noise and interference variance and SNR.

Making use of simple OFDM model of section 2.2.4, we can write

$$rn[i] = hn[i].sn[i] + vn[i];\ and\ rn[j] = hn[j].sn[j] + vn[j] \quad (4.8)$$

$$\Delta n = rn[i] - rn[j] = vn[i] - vn[j] \quad (4.9)$$

$$\sigma_{N+1}^{2} = \frac{1}{2}E\{|\Delta_n|^2\} \quad (4.10)$$

Assumptions

- This algorithm assumes transmission of two identical sequences to be sent towards the receiver.
- It assumes channel to remain invariant during the reception of these two identical sequences.
- It assumes that same noise prevails over N symbols of identical sequences so that in the end it averages out over N symbols.

Advantages

- Channel estimation errors don't affect the SNR estimation.

Disadvantages

- If intentionally identical sequences are transmitted, it will have a bad impact over DL throughput otherwise UE needs to wait for an unknown time until it receives two identical sequences. If we make UE wait then assumption of this algorithm that channel remains same during the reception of two identical sequences goes weaker.
- This algorithm is highly dependent upon the number of interferers.

This algorithm does not seem suitable for our requirement of estimation of instantaneous SINR especially because it requires transmission of two identical sequences which will degrade DL throughput. On the other hand this algorithm will not give good results if system is running in a fast fading channel.

4.3.3 Moment Based Blind Algorithm (M2M4)

As this algorithm is blind, it does not require channel estimation or symbol detection. It directly works over the received symbols. It calculates the 2^{nd} order and 4th order moments of the received signal and then using the knowledge of kurtosis for data, noise and channel, it succeeds in calculating the noise variance and then it calculates SNR estimate.

Assumptions

- It assumes that same noise prevails over N symbols so that in the end it averages out over N symbols.

Advantages

- Channel estimation errors don't affect the SNR estimation.
- Symbol detection errors don't affect the SNR estimation.

Disadvantages

- M2M4 algorithms is valid for the very large data sequences which undergo large channel variations which is not at all the case for our problem.
- This algorithm is not valid for M-ary PSK in Rayleigh Fading Channel because in that case 2nd and 4th order moments don't give independent information and one can be directly written in the form of other.

The aspect that this algorithm is not applicable to QPSK, the basic transmission technique of OFDMA, and requires large channel variations make it unsuitable to select. And as it is totally blind and does not use channel estimates or detected data, if we have any of these available, an algorithm which uses these information will give better results as compared to M2M4.

4.3.4 Signal Regeneration Method

In Signal Regeneration algorithm, receiver receives a symbol and having done the estimation of channel and detection of received symbol, it builds a noiseless regenerated signal which is subtracted from the received signal to obtain an evaluation of the additive noise in the received symbol. Thus accuracy of this method depends upon the reliability of channel estimation and symbol detection.

$$v_n = r_n - \hat{h}_n . \hat{s}_n \tag{4.11}$$

$$= \left(h_n . s_n - \hat{h}_n . \hat{s}_n\right) + v_n \tag{4.12}$$

$$\sigma_{N+1}^2 = E[|\hat{v}_n|^2] \tag{4.13}$$

Thus in this algorithm, statistical average is converted to temporal average.

Assumptions

- This algorithm works over the implicit assumption that channel estimation and symbol detection are nearly perfect.

- It assumes that same noise prevails over N symbols of this chunk so that it averages out over N symbols.

Advantages

- This method is applicable to data symbols and pilot symbols both and even by using the pilot symbols, its performance goes better because then the effect of symbol detection errors is removed, as pilots are known symbols.

Disadvantages

- This algorithm is free of any particular kind of limitations but its performance depends upon the accuracy of channel estimation and symbol detection.

This algorithm seems highly suitable to be utilised in our SINR estimation for OFDMA chunks. Its added advantage is the fact that it is applicable to both data symbols and pilot symbols so for user's assigned chunk it can be used for SINR estimation using both kinds and for other chunks not allocated to this particular UE, this algorithm has the capability of giving SINR estimate by having applied over pilot symbols only or their data can be hard detected to use this method.

4.3.5 Moment Based SINR Estimation Algorithm for Rayleigh Fading Channel

This algorithm for SINR estimation makes use of both the 2nd order auto correlation and 2nd order cross correlation of the received signal. Auto correlation of the received signal gives UE an estimate of combined power of signal and noise. This algorithm calculates cross correlation of signal with unit delay and with delay of 2 symbols. A particular manipulation of these cross correlation values, using the fact that correlation of Rayleigh fading channel is 0th order Bessel Function of 1st kind, gives us signal power. So it manages to calculate signal power and noise power from which it gets SINR estimation.

Assumptions

- This algorithm is a pilot based algorithm so symbols should be known to the receiver.

- It goes further in assuming that same pilot symbols are sent over the same sub-carriers.

- It assumes that same noise prevails over N symbols of sub-frame so it averages out over N symbols.

Advantages

- This algorithm does not use the channel estimates, so is independent of the channel estimation errors, which are normally not negligible.

- As compared to M2M4 moment based algorithm, it estimates SINR just from 2nd order statistics and does not go for 4th order moments etc.

Disadvantages

- This algorithm is valid only for pilot symbols and even when same pilot symbol is carried by same sub-carrier.

- In the calculation of cross correlation expression, it makes approximations in the series expansion of Bessel Function. It needs to be explored the impact of this approximation over the SINR estimation. Worse than this is the fact that representation of correlation of Rayleigh fading channel as Bessel Function holds for Omni-directional antennas. But in practical networks, we always have directional antennas.

As we saw that this algorithm is valid only when same pilot symbols are used over sub-carriers and only with omni-directional antennas which make this algorithm unusable in practical situations. So we will avoid this algorithm.

4.3.6 Sliding Window Algorithm

This algorithm does SINR estimation by taking into account the colour of noise, so instead of averaging the instantaneous noise sample estimates over all the sub-carriers of one chunk, it divides the sub-carriers into several sub groups and averaging of SINR over the sub carriers separately within each sub group is proposed. Also these averaged estimates over each sub group are further averaged across several OFDM symbols in time to enhance the performance of the noise power estimates. Simply it creates frequency time rectangular windows for averaging out the noise power estimates and window-lengths in time and frequency, of course, depend upon the colour of noise.

Assumptions

- This algorithm makes use of Signal Regeneration Algorithm as a core algorithm for SINR estimation but over this core it makes some significant changes.
- One of the most important features of this algorithm is that it removes the assumption of white noise over all sub-carriers of a particular chunk which makes this algorithm unique from others.

Advantages

- It will give us more accurate information of SINR for each chunk.
- This algorithm is flexible in the sense that for SINR estimation we can use any core SINR estimation algorithm of our own choice.

Disadvantages

- The fact is clear to us that number of data symbols in one chunk, are hardly sufficient to make good estimation of SINR. This algorithms further divides one chunk in multiple T-F windows, which will obviously have a very small number of data symbols, making it almost impossible to have good estimation of SINR over them.

- This algorithm remarks that T-F window lengths should depend upon the colour of noise but gives no concrete details about calculating these parameters. Channel coherence bandwidth and channel coherence time may help us in this regard to identify the colour of noise but if interference also varies over one chunk, it would be really difficult to make T-F windows depending upon the colour of interference.

This algorithm can be used as a wrapper over some basic SINR estimation algorithm. Because this unique idea of multiple SINR values for one chunk depending upon the colour of noise is important.

4.4 Choice of Algorithms for SINR Estimation

In section 4.3, we presented the state of the art algorithms which exist in literature for estimation of noise variance or SINR. In the same section, all these algorithms were analyzed critically, mentioning the assumptions they make, giving advantages and disadvantages of each. The analysis of Signal Regeneration Algorithm in section 4.3.4 indicated to us that this algorithm may be used in our case of estimation of instantaneous SINR.

As channel estimates are always provided by channel estimator, so whenever data detection is possible (soft or hard), SR method becomes the best choice to use.

4.5 Numerical Results

4.5.1 System Parameters

Almost all of the system parameters in our simulation environment are mentioned in the following table:

Parameter Name	Parameter Value
System Bandwidth	10 MHz
Sub-Carrier Spacing	15 KHz
Sampling Frequency	15.36 MHz
FFT Size	1024
Number of Sub-Carriers	600
Guard Interval	72 Samples
Size of Sub-Frame	7 OFDM Symbols
Duration of Sub-Frame	0.5 ms
Sub-Carriers per PRB	25
Transmission Mode	Localized
Modulation Type	QPSK (Gray Mapping)

4.5.2 Transmission Modes

The following graphs show estimated SINR vs actual SINR and Root Mean Square Error (RMSE) between estimated SINR and actual SINR for four modulation schemes i-e BPSK,QPSK,16-QAM and 64-QAM in both transmission modes (Localized / Distributed).

1. Localized

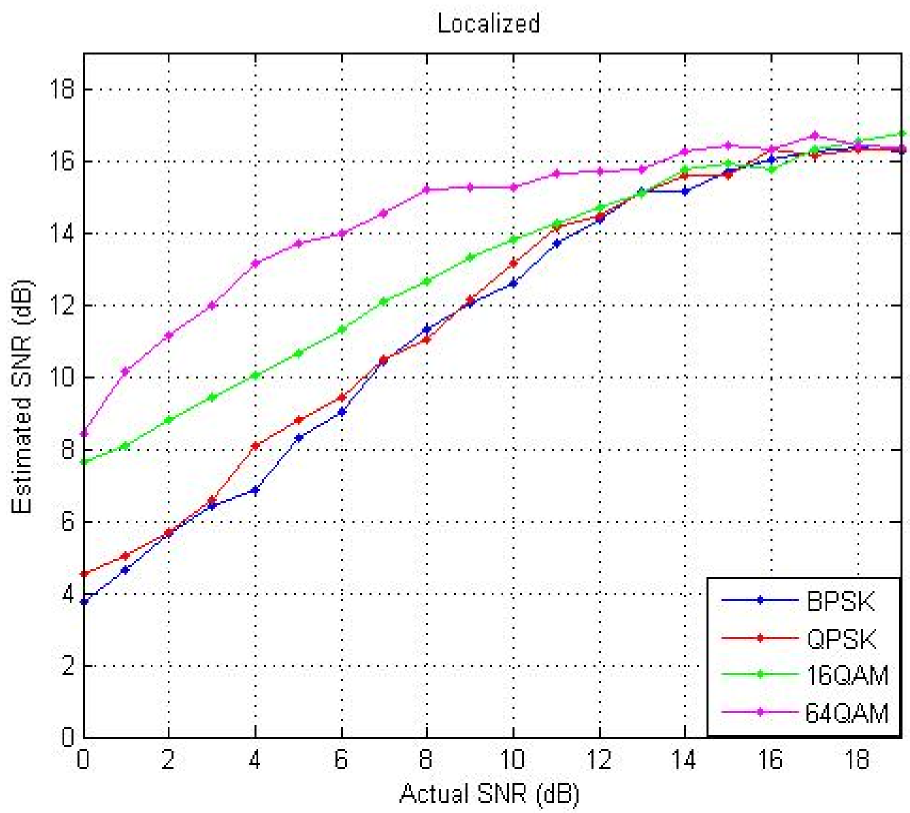

Figure 4.1: Relationship between Actual & Estimated SINR in Localized Mode

The graph shows that as we move from lower SINR to higher value of SINR, the estimated value comes closer to the actual one. In terms of modulation schemes, BPSK shows the best results among these four schemes while 64QAM shows the other way round.

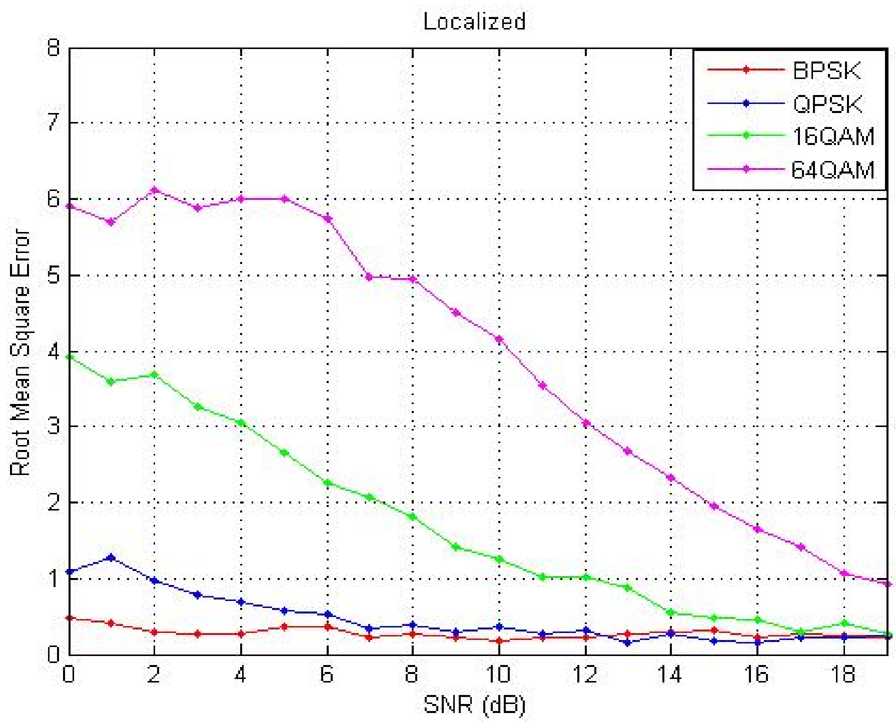

Figure 4.2: RMSE against Actual SINR in Localized Mode

The graph shows that as we move from lower SINR to higher value of SINR, the value of RMSE reduces to less than 1 db. As far as modulation schemes are concerned, BPSK has the least error while 64 QAM bears maximum when compared with other modulation schemes.

2. Distributed

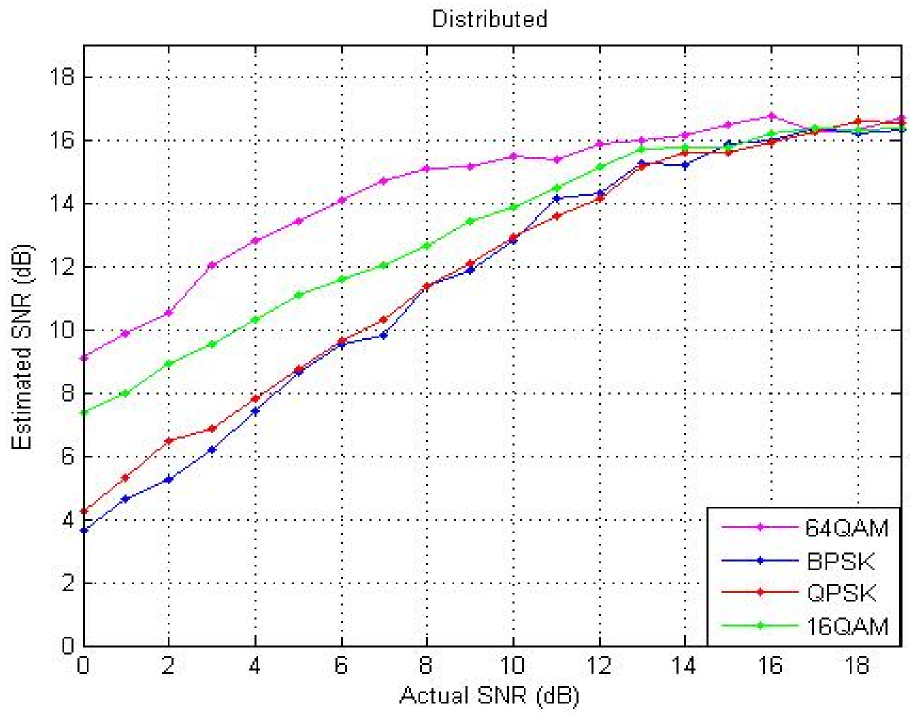

Figure 4.3: Relationship between Actual & Estimated SINR in Distributed Mode

The above graph shows that as we go from lower SINR to higher value of SINR, the estimated value comes closer to the actual one. In terms of modulation schemes, BPSK shows the best results among these four schemes while 64QAM shows the other way round.

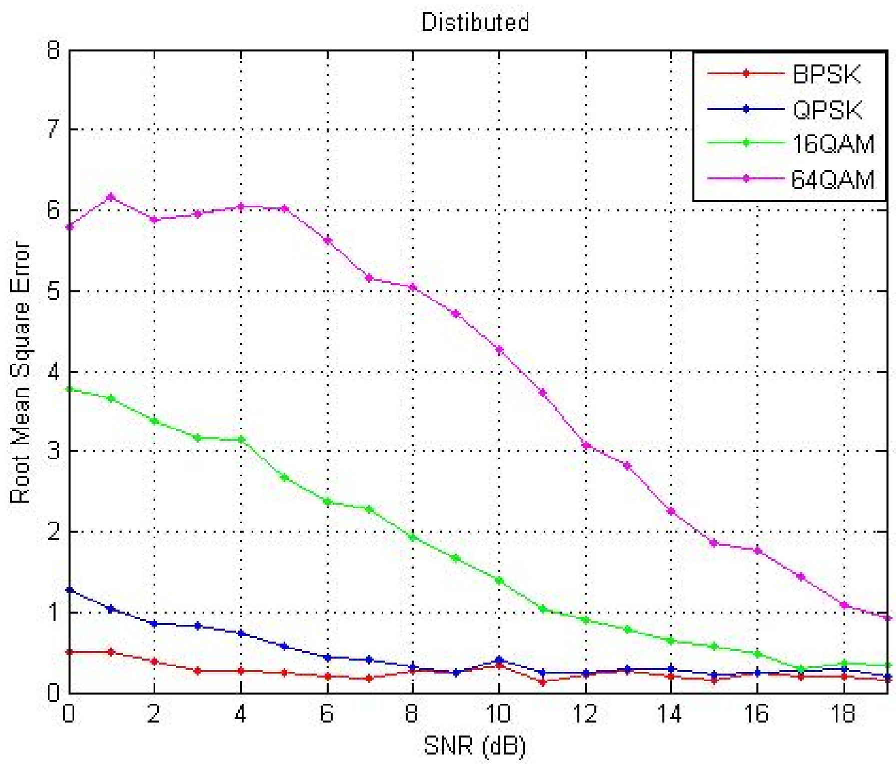

Figure 4.4: RMSE against Actual SINR in Distributed Mode

The graph given above shows that as we go from lower SINR to higher value of SINR, the value of RMSE reduces to less than 1 db. As far as modulation schemes are concerned, BPSK has the least error while 64 QAM bears maximum when compared with other modulation schemes.

4.6 Conclusions

This chapter focuses over the CQI measurements which are necessary to be made at user's side and return to BS so that BS can take suitable decisions for the tasks associated with link adaptation (scheduling, AMC and HARQ), and for intercell interference avoidance. Thus we show that effective SINR (geometric mean of SINR over symbols of a sub-frame) may work as CQI with the help of look-up tables. We present the state of the art SINR estimation algorithms and choose SR and M2 for our use. In the end, numerical results have been presented for estimation of SINR. The graphs showing RMSE in the estimation of SINR show that SINR can be estimated with sufficient accuracy so when used as feedback, it can provide us considerable gains because of link adaptation processes and inter-cell interference avoidance.

Conclusions

The main challenge for future wireless communication systems is to achieve the high data rates, give good performance for a broad range of applications, maintaining QoS requirements at all times, and on the cost of minimal utilization of resources. Link adaptation is a term widely used in wireless communications to denote the matching of the modulation, coding and other signal and protocol parameters to the quality of transmission. Our system of interest for this thesis project has been OFDMA. OFDMA is highly probable candidate for the DL of future wireless radio access networks. OFDMA enjoys frequency reuse factor of 1, removing all the issues of frequency planning. But it will combat inter-cell interference employing frequency hoping. The decisions, required for implementing frequency hoping and all the processes of link adaptation at the transmitter, demand some kind of active feedback from the user. This feedback should reflect the quality of transmission in terms of channel quality and interference conditions. The measurements for this feedback, called CQI feedback, have been the subject of this thesis project.

To present all of the things in a systematic manner, we presented good details for system of concern, OFDMA and about multi-path channel in the beginning. Later from the existing literature, we showed points of interest for link adaptation processes. All of these (frequency scheduling, AMC and HARQ) may play a very important role for achieving high data rates maintaining QoS parameters, the objectives for next generation networks. But as we already mentioned that these processes require user's feedback about multiple chunks in OFDMA sub-frame. This feedback should be function of some metric, user can measure or estimate. Thus we have showed that effective SINR over one chunk may work as this metric which can represent CQI. This effective SINR is geometric mean of SINR over symbols of this chunk. So we show our problem of CQI measurements transforming towards estimation of SINR.

For our problem of SINR estimation, we showed that theoretically Signal Regeneration algorithm is the optimal choice for our system of interest. We suggest to always employ data symbols for this estimation. Results presented for estimation of

SINR prove that estimation of SINR is possible with sufficient accuracy and thus when given as feedback to BS transmitter, it may have a good idea of channel transmission quality. In other words it proves that all the gains promised by link adaptation techniques and by interference avoidance are quite realizable in this system of DL OFDMA.

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