BASICS OF CDMA AND OFDM, DIVERSITY AND MIMO EQUALIZATION

CDMA: Code Division Multiple Access. ***Abstract*****—***Code Division Multiple Access (CDMA) is probably the most interesting multiple access method provided by spread spectrum technology. CDMA refers to a multiple access method in which the individual terminals use spread spectrum techniques and occupy the entire spectrum whenever they transmit. This lecture focuses on direct-sequence spread-spectrum principles (processing gain, spreading, de-spreading, modulation and codes) and more into CDMA cellular system fundamentals like multiple access interference (MAI), multipaths, RAKE receiver, power control and soft handover. Also, the capacity of a CDMA system is discussed.*

INTRODUCTION

CDMA refers to a multiple access method in which the individual terminals use spread-spectrum techniques and occupy the entire spectrum whenever they transmit. This feature makes CDMA different from frequency division multiple access (FDMA) and from time division multiple access (TDMA). In FDMA each user is given a small portion of the total available spectrum, and in TDMA each user is allowed full use of the available spectrum, but only during certain periods in time. Code Division Multiple Access is a modulation and multiple access scheme based on the spread-spectrum communication technology. It is well-established technology and applied to digital cellular radio and wireless communication systems in the early 1990s. Capacity concerns of major markets and efficient and economic wireless communication needs of the industries were the most significant drivers for the development of the CDMA cellular technology. CDMA is a method in which users share time and frequency allocations, and are channelized by unique assigned codes. The signals of different users are separated at the receiver by using a correlator that captures signal energy only from the desired user or channel. Undesired signals contribute only to noise and interference. Figure 1 illustrates the principle of the CDMA technique.

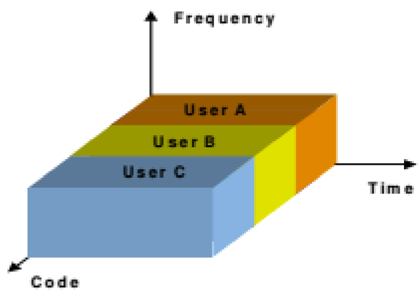

Figure 1. CDMA multiple access method

The development of the CDMA technique dates back to the early 1950s when different studies of the spread-spectrum technologies were started. The first era in CDMA history consisted of introducing basic ideas of the CDMA by Claude Shannon and Robert Pierce in 1949. In 1950 De-Rosa-Rogoff defined the direct-sequence spread-spectrum method, the processing gain equation, and a noise multiplexing idea. Price and Green filed the RAKE receiver patent in 1956. In 1961 Manuski defined the near-far problem crucial for CDMA systems. During the 1970s several military and navigation applications were developed. The second CDMA era introduced studies focusing on narrowband systems. In 1978 Cooper and Nettleton suggested a cellular spread-spectrum application. During the 1980s communication company Qualcomm investigated narrowband CDMA techniques for cellular applications, and the result was that in 1993 the CDMA IS-95 standard was developed. Compared to third generation CDMA systems IS-95 can be considered a narrowband CDMA system with 1.2288 Mchip/s carrier chip rate. Third generation wideband CDMA systems, such as CDMA IS-2000 and European WCDMA use higher chip rates than CDMA IS-95.

SPREAD-SPECTRUM TECHNOLOGY

Originally, the spread-spectrum technology has been developed for military and navigation purposes because it has some interesting characteristics that provide secure means of communication in hostile environments [2]. First of all, spread spectrum signals have LPI-properties (Low Probability of Interception), and cannot be easily detected by enemy communication equipment due to low power spectral density, even lower than background noise. Secondly, spread spectrum signals have efficient AJ (Anti-Jamming) properties to combat intentional interference trying to sabotage communication systems. Nowadays, spread-spectrum technology has also proven to be feasible for commercial applications especially for mobile communication systems. It provides an efficient multiple access method for a number of independent users sharing a common communication channel without external synchronization methods. CDMA is probably the most interesting multiple access method provided by spread-spectrum technology. The fundamental idea of spread-spectrum communication is to spread a certain information bandwidth, Bi, over a wider transmission bandwidth, Bt. The minimum of the transmission bandwidth has to be wider than the information bandwidth. The relative rate between user information and the pseudo-random code sequence can be on the order of tens or hundreds for commercial systems and on the order of thousands for military systems. Spread-spectrum communications cannot be said to be an efficient means of utilizing bandwidth because it needs a lot of bandwidth to be efficient. On the other hand, the wider transmitted bandwidth offers such a low power spectral density that it makes the transmitted signal look like background noise in front end of a receiver. Besides LPI and AJ capabilities spread spectrum communication systems can offer further advantages such as multiple access, efficient privacy, and interference rejection. There are two basic spread-spectrum techniques: direct sequencing (DS), frequency hopping (FH), and time hopping (TH). Also, a variety of hybrid techniques use different combinations of these basic techniques. With direct-sequence spreading, the original signal is multiplied by a known signal of much larger bandwidth. With frequency-hopped spreading, the center frequency of the transmitted signal is varied in a pseudorandom pattern.

Processing Gain

Combining a bit stream of information with an independent pseudo-random code sequence by simple multiplication carries out the spreading operation. One of the main parameters of a spread spectrum communication system is the processing gain, Gp. It is the ratio of the transmitted bandwidth, Bt, and information bandwidth, Bi, as presented in the following equation 1.

$$G_p = \frac{B_t}{B_i} \qquad (1)$$

Gp is also called the spreading factor. This processing gain or spreading factor determines the maximum number of simultaneous users or connections allowed in a communication system. It determines the level of protection against multipath interference signals and signal detection capabilities of a spread spectrum communication system. In multipath situations the receiver observes spread spectrum signals summed with narrowband interference. The processing gain determines the power ratio of the desired signal and interference after de-spreading. Higher desired signal power leads to easier detection. It can be seen that low data rates such as speech have high processing gain compared to high data rates.

DIRECT-SEQUENCE CDMA

As mentioned CDMA is a spread-spectrum multiple access method. Spread-spectrum is a transmission method in which the signal occupies a bandwidth in excess of the minimum necessary to send the information. The spreading of the signal is accomplished by means of a pseudorandom code that is independent of the transmitted data signal. A synchronized reception with the same pseudorandom code at the receiver is used for de-spreading and subsequent data recovery.

Principle of DS-CDMA

Figure 2 shows the multiple access capability of a CDMA communication system. Two users are sending simultaneously narrowband information signals having the same bandwidth Bi. Both narrowband signals are spread with a user specific and unique code having sufficiently low cross correlation with the other user’s code. Code makes each user’s communications approximately orthogonal to those of other users. After spreading the two signals are transmitted into a radio channel having the same bandwidth, Bt. In the radio channel the two signals are mixed and exposed to impairments. Spreading the signal de-sensitizes the original narrowband signal to some potential channel degradation and to interference. The signals cannot be distinguished from each other and from background noise due to their low powers achieved by the spreading. The transmitted energy remains the same, but due to much larger bandwidth, the signal spectrum is often below the noise floor of receivers. At the receiver the desired narrow-band information signal can be extracted or de-spread by a replica of the spreading code used in transmitter for a particular user. Signals for other users are not de-spread; they are spread more.

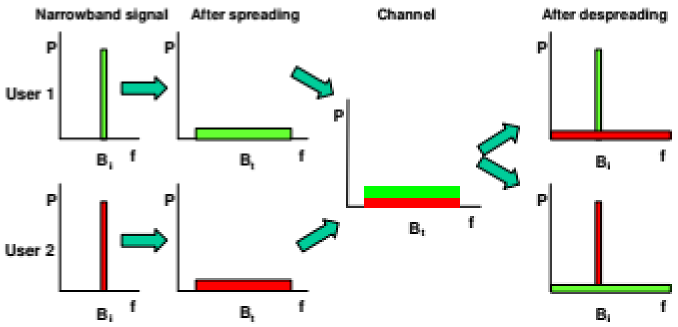

Figure 2. CDMA multiple access principle

DS-CDMA Transmitter and Receiver

The basic DS-CDMA transmission and reception is illustrated in Figure 3.

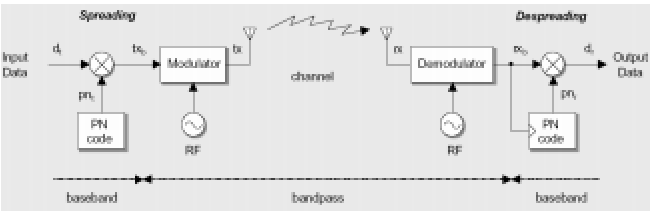

Figure 3. Transmission and reception in DS-CDMA

Signal transmission consists of the following steps. A pseudo-random code is generated, different for each channel and each successive connection. The information data is spread by pseudorandom code. The resulting signal modulates a carrier. The modulated carrier is amplified and broadcast. Signal reception consists of the following steps. The carrier is received and amplified. The received signal is mixed with a local carrier to recover the spread digital signal. A pseudorandom code is generated, matching the anticipated signal. The receiver acquires the received code and phase locks its own code to it. The received signal is correlated with the generated code, extracting the information data.

Spreading

In the transmitter, the binary data is directly multiplied (XOR-function) with the pseudo-noise sequence, which is independent of the binary data, to produce the transmitted baseband signal having much wider bandwidth than the original signal. This is presented in **Figure 4**.

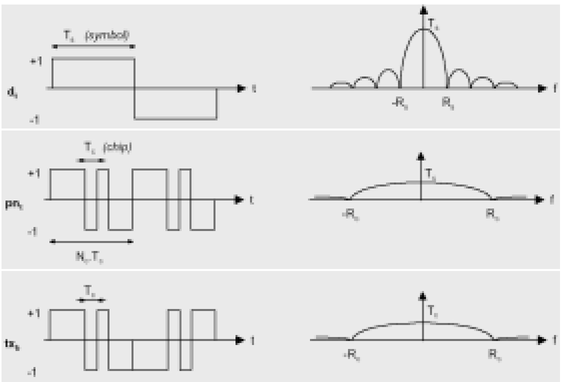

Figure 4. Spreading in DS-CDMA

De-spreading

In the receiver the baseband signal is multiplied with the same pseudo-noise sequence. If the pseudorandom code is not the same or it is not in synchronization with the data there is no de-spreading.

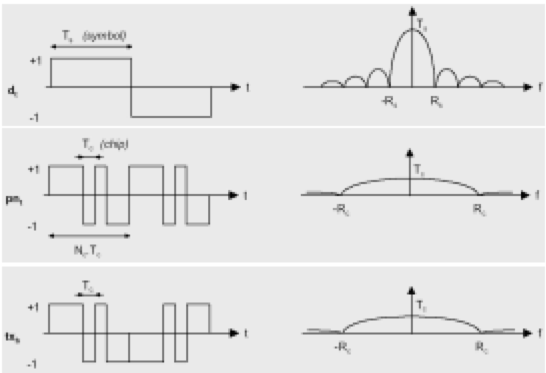

Figure 5. De-spreading in DS-CDMA

DS Modulation

Spread-spectrum techniques can be used with many modulations formats, but most practical applications are limited to BPSK (Binary Phase Shift Keying) and QPSK (Quadrature Phase Shift Keying).
Figure 6 shows an example of the generation of a BPSKmodulated spread-spectrum signal. The basic idea is that the signal after the spreading operation i.e. the multiplication is 1 if the two signals are the same, 1 or 0. Otherwise the output is 0. The BPSK-signal has a 180-degree phase shift when the output of the multiplication changes.

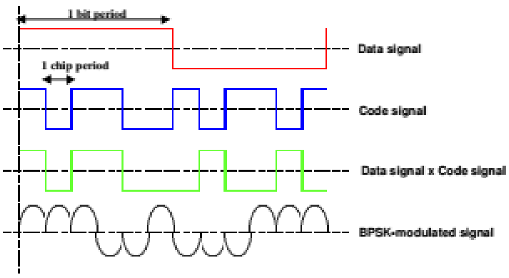

Figure 6. Generation of BPSK-modulated spread spectrum signal

One advantage of DS modulation is the reduced receiver sensitivity to interference. This advantage is due to the fact that the de-spreading circuit acts as a spreading circuit for any signal to which it is not matched.

Pseudo-Noise Sequences

The DS-CDMA system uses two general categories of spreading sequences: PN sequences and orthogonal codes. The PN sequence is produced by the pseudo-random noise generator that is simply a binary linear feedback shift register, consisting of XOR gates and a shift register. This PN generator has the ability to create an identical sequence for both the transmitter and the receiver, and yet retaining the desirable properties of a noise-like randomness bit sequence.

A PN sequence has many characteristics such as having a nearly equal number of zeros and ones, very low correlation between shifted versions of the sequence and very low cross correlation with any other signals such as interference and noise. However, it is able to correlate very well with itself and its inverse. Another important aspect is the autocorrelation of the sequence as it decides the ability to synchronize and lock the spreading code to the received signal.

CDMA IN CELLULAR ENVIRONMENT

CDMA is a multiple access for wireless communications based on direct-sequence spread-spectrum. All users can transmit at the same time, and each is allocated the entire available frequency spectrum for transmission. CDMA does not require the bandwidth allocation of FDMA, nor the time synchronization of the individual users needed in TDMA. A CDMA user has full time and full bandwidth available, but the quality of the communication decreases with an increasing number of users.

Multiple Access Interference (MAI)

The detector receives a signal composed of the sum of all users' signals, which overlap in time and frequency. MAI refers to the interference between DS users and is a factor, which limits the capacity and performance of the system.

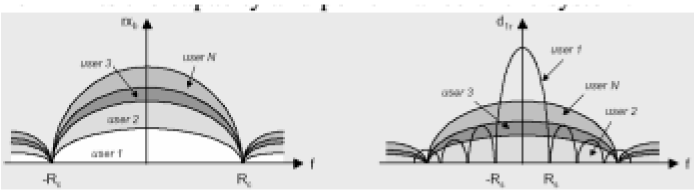

Figure 8. MAI in CDMA

With CDMA systems, the same frequency channel can be used in the adjacent cell, as long as multiple access interference is kept below a given level. This MAI is directly proportional to the channel loading. MAI can be divided in two parts: intra-cell and inter-cell interference.

B. Multipath Channel

Multipath is the reception of multiple, possibly interfering copies of the same signal. The tolerance of spread-spectrum techniques to interference extends also to a tolerance of multipath. With some receiver designs, multipath can even be used advantageously.

RAKE Receiver

Due to reflections from obstacles a wideband radio channel can consist of many copies (multipaths) of originally transmitted signals having different amplitudes, phases, and delays. If the signal components arrive more than duration of one chip apart from each other, a RAKE receiver can be used to resolve and combine them. The RAKE receiver uses a multipath diversity principle. It is like a rake that rakes the energy from the multipath propagated signal components. When a wideband signal is received in a matched filter over a multipath channel, the multiple delays appear at the receiver, as depicted in Figure 10. The RAKE receiver uses several baseband correlators to individually process several signal multipath components. The correlator outputs are combined to achieve improved communications reliability and performance.

Impulse response measurements of the multipath channel profile are executed through a matched filter to make a successful de-spreading. It reveals multipath channel peaks and gives timing and RAKE finger allocations to different receiver blocks. Later it tracks and monitors these peaks with a measurement rate depending on speeds of mobile station and on propagation environment. The number of available RAKE fingers depends on the channel profile and the chip rate. The higher the chip rate, the more resolvable paths there are, but higher chip rate will cause wider bandwidth. To catch all the energy from the channel more RAKE fingers are needed. A very large number of fingers lead to combining losses and practical implementation problems.

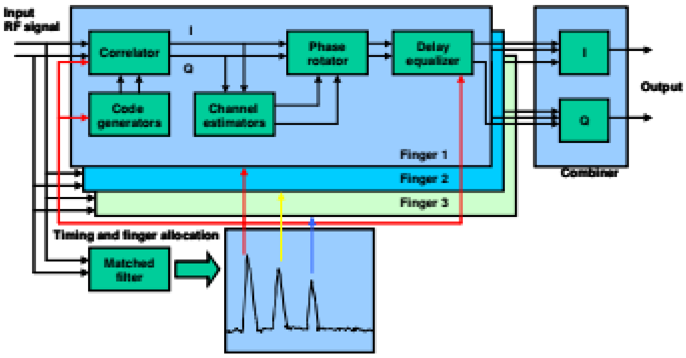

Figure 9. Block diagram of simple RAKE diversity receiver

Near-far Problem

Figure 10 illustrates the near-far problem associated with CDMA-based systems. The problem arises when MS A (Mobile Station) and MS B are located in a same cell with different distances from a BS (Base Station). If no power control were applied in uplink, the MS A would transmit so high power that MS B would have no connection to the BS due to too low SIR-values. The MS A would be reserving a great amount of the capacity of the cell. Power control is implemented to overcome the near-far problem and to maximize the capacity of the system [3]. It tries to control the powers of the mobile stations in the system so that the received powers at the base station stay equal. It tries also to compensate the effects of slow fading and fast fading. There is no near-far problem in downlink due to a one-to-many situation. All the signals within one cell originate from the one base station to all mobiles.

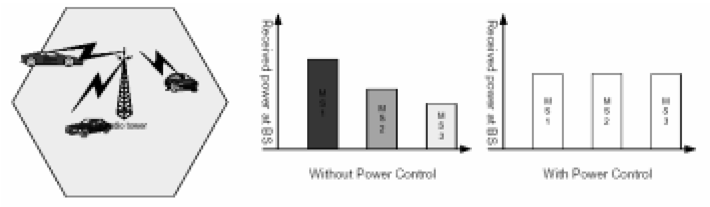

Figure 10. Near-far problem

E. Power Control

Power control is an extremely essential function when considering the smooth operation and the capacity of CDMA based systems. The power control problem arises due to multiple access interference. Each user looks like random noise to other users and causes unnecessary interference to the system. That is why the powers of individual users have to be carefully controlled. Power control forces all users to transmit the minimum amount of power needed to achieve acceptable signal quality at the base station. Typically, it reduces the power transmitted by the users closest to the base station, while increasing the power of the users farthest away from the base station. Power control tries to set the power received from all users to be equal at the base station receiver. A secondary reason for power control is to minimize battery consumption in the mobile.

Soft and Softer Handover

This type of handover is characterized by commencing communications with a new base station on the same CDMA frequency assignment before terminating communications with the old base station. Softer handover occurs between two or more cells of one base station. A soft and softer handover prevent the ping-pong behavior, and the dual base station capability is a form of diversity that can increase capacity in a heavily loaded system and also coverage in a lightly loaded system. Soft handover helps to minimize with power control the interference both in uplink and downlink directions.

CDMA CAPACITY

In a CDMA-based system capacity can be defined as throughput of bits or as the amount of simultaneous users in the network receiving voice and data services with certain predefined quality targets. Interfering signals caused by users to each other rise, as the amount of users gets higher in the network. A balance between maintaining connection integrity and restricting interference level is maintained by controlling the power of each user so that signals arrive at their intended receiver with minimum required S/N-level. Interference, coverage, and capacity are coupled tightly to together in a CDMA system. Capacity can be restricted by either transmission power constraints or by the self-generated interference. In the uplink, the system reaches its capacity when a mobile station does not have enough power to overcome interference from the network, or a predefined loading target of the network is met. In the downlink, capacity is reached when no additional power is available to add new users i.e. base station transmission power restrictions are met, or there are no downlink codes available for mobile stations. The power needed for either link is fundamentally related to Eb/No requirements for different services.

CDMA CELLULAR APPLICATIONS

Cellular wireless communications have evolved from first generation analog techniques to the more flexible second and third generation digital techniques that are currently employed. Future developments are aimed at further

CONCLUSION

This lecture has introduced direct-sequence spread-spectrum basics and pointed out some important CDMA cellular requirements on DS-SS. DS-CDMA is probably the most interesting multiple access method provided by spread spectrum technology. In DS-CDMA data is scrambled by user specific pseudo-noise code at the transmitter. The effect of interference can be reduced by the processing gain. Through RAKE receiver multipaths can be used in advantage to improve receiver performance by capturing the energy in paths having different transmission delays. In fading channels, by use of the RAKE receiver, a SS receiver can obtain an important advantage in diversity. When considering cellular CDMA system some aspects are really important. Power control and soft handover must work or there is no cellular CDMA as we know. Interference, coverage, and capacity are coupled tightly to together in CDMA systems. Nowadays systems such as CDMA2000 and its evolution versions, and European WCDMA are becoming more and more popular as the networks are opening commercially around the world. The third CDMA era is now starting.

OFDM: Orthogonal Frequency Division Multiplexing and its application. **Abstract*:*** ***Orthogonal Frequency Division Multiplexing (OFDM) is a multi-carrier modulation technique which is very much popular in new wireless networks of IEEE standard, digital television, audio broadcasting and 4G mobile communications. The main benefit of OFDM over single-carrier schemes is its ability to cope with severe channel conditions without complex equalization filters. It has improved the quality of long-distance communication by eliminating Inter Symbol Interference (ISI) and improving Signal-to-Noise ratio (SNR). The main drawbacks of OFDM are its high peak to average power ratio and its sensitivity to phase noise and frequency offset. This second part of the lecture gives an overview of OFDM, its applications in various systems such as IEEE 802.11a, Digital Audio Broadcasting (DAB) and Digital Broadcast Services to Handheld Devices (DVB-H) along with its advantages and disadvantages.***

Introduction

During the last few decades, growth rate of wireless technology has been accelerated to such a level that it has become ubiquitous. Progress in fiber-optics with assurance of almost limitless bandwidth and predictions of universal high-speed wireless internet access in the not-too-distant future thrive in both the popular press and technical journals. Wireless communication is having the fastest growth phase in history because of unprecedented evolution in the field. The kid of wireless communication is experiencing golden days due to various wireless standards such as Wi-Fi, GSM, Wimax and LTE. These standards operate within lower microwave range (2-4GHz). Due to intrinsic propagation losses at these frequencies and problem of multipath fading, it was necessary to provide a solution which can offer robustness in multipath environments and against narrowband interference and is efficient. OFDM, in all this aspect, proves to be an apt candidate by not only providing high-capacity, high-speed wireless broadband multimedia networks but also coexists with current and future systems. Orthogonal frequency-division multiplexing (OFDM) is a method of digital modulation in which a signal is split into several narrowband channels at different frequencies. OFDM has been adopted by several technologies such as Asymmetric Digital Subscriber Line (ADSL) services, IEEE 802.11a/g, IEEE 802.16a, Digital Audio Broadcast (DAB), and digital terrestrial television broadcast: DVD in Europe, ISDB in Japan 4G, IEEE 802.11n, IEEE 802.16, and IEEE 802.20. OFDM converts a frequency-selective channel into a parallel collection of frequency flat sub channels. Though it is derived from frequency division multiplexing (FDM), OFDM provides many advantages over this conventional technique. In OFDM the subcarrier frequencies are chosen so that the signals are mathematically orthogonal over one OFDM symbol period. Both modulation and multiplexing are attained digitally using an inverse fast Fourier transform (IFFT) and thus, the required orthogonal signals can be generated accurately.

Architecture of OFDM

Practically, OFDM modulation for standard IEEE 802.20 is used by both the forward and reverse links. IEEE 802.20, also referred to as Mobile-Fi, is optimized for IP and roaming in high-speed mobile environments. This standard is ready to fully mobilize IP, opening up major new data markets beyond the more circuit-centric 2.5G and 3G cellular standards. Its main operation is to develop the specification for an efficient packet-based air interface optimized for the transport of IP-based services. For IEEE 802.20, transmission on the forward link is divided into super frames, where each super frame consists of a preamble followed by a sequence of 25 Forward Link Physical Layer (NFLPHY) frames. Transmission on the reverse link is also divided into units of super frames, with each super frame consisting of a sequence of 25 reverse link PHY frames. In order to support cell sizes of macro, micro, and pico IEEE 802.20 should operate in a traditional cellular environment. To increase the availability of coverage area, increase throughput available to the users, and enable a higher overall spectral efficiency, advanced antenna technologies such as multi antenna at the base station should be employed. The mathematical description for the OFDM signal is given as follows: The low-pass equivalent OFDM signal is expressed as

$$X(t) = \sum_{k=0}^{N-1} X_k e^{j2\pi kt/T} \qquad , 0 \le t < T \qquad (1)$$

this is also Discrete Fourier Transform (DFT). Here Xk are data symbols which is sequence of complex numbers representing BPSK, QPSK or QAM baseband symbol, N is number of subcarriers and T is OFDM symbol line. The subcarrier spacing 1/t makes them orthogonal over each symbol period. Sequence of OFDM symbols is given as follows:

$$S(t) = \sum_{k=-\infty}^{+\infty} X(t - kT) \qquad (2)$$

To avoid ISI, a guard interval of length *Tg* is inserted before OFDM block. During this interval, a cyclic prefix is transmitted. The signal with cyclic prefix is thus given as,

$$X(t) = \sum_{k=0}^{N-1} X_k e^{j2\pi kt/T} \qquad , -T_g \le t < T \qquad (3)$$

The Fast Fourier Transform (FFT) is a computationally efficient implementation of DFT. Inverse Fast Fourier Transform (IFFT) and FFT are main modulation and demodulation techniques used in OFDM.

Application of OFDM in various systems
In Standard IEEE 802.11a

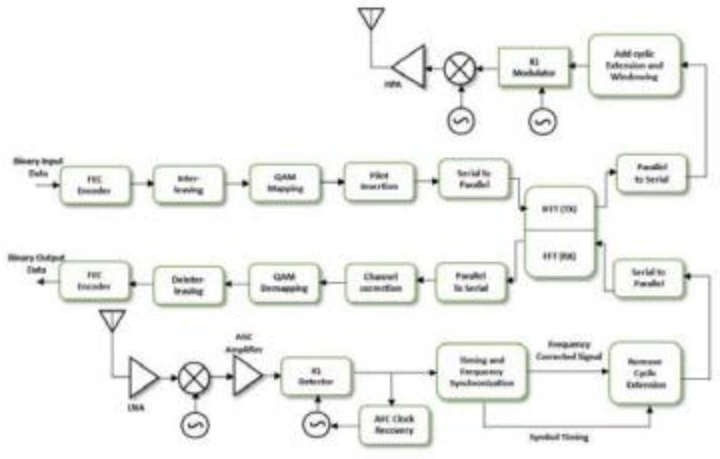

Figure 1: Block diagram of OFDM transceiver [6]

In the transmitter, input data which is in binary is encoded by a rate half convolution encoder. After interleaving, the binary values are converted to QAM values. Four pilot values are added to each 48 data value, so that coherency at the reception point can be achieved. It gives 52 QAM values per OFDM symbol. Application of IFFT modulates the symbol onto 52 subcarriers. Cyclic prefix is added to make the system robust to multipath propagation. Narrower output spectrum is obtained by applying windowing. Using an IQ modulator, the signal is converted to analog, which is up converted to the 5 GHz band, amplified, and transmitted through the antenna. The receiver performs the reverse operations of the transmitter, with few additional tasks. In the first step, the receiver has to estimate frequency offset and symbol timing, using special training symbols in the preamble. After removing the cyclic prefix, the signal can be applied to a Fast Fourier Transform to recover the 52 QAM values of all subcarriers. The training symbols and the pilot subcarriers are used to correct for the channel response as well as remaining phase drift. The QAM values are then demapped into binary, and finally a Viterbi decoder decodes the information bits. Fig. 1 depicts block diagram of OFDM transceiver.

In Digital Audio Broadcasting (DAB)

Digital Audio Broadcasting (DAB) is a digital radio technology for broadcasting radio stations, used in several countries, especially in Europe. It has 4 transmission modes with different parameters as shown in the Table 1.

Table 1: Digital Audio Broadcasting parameters [7]

Parameters	Mode I	Mode II	Mode III	Mode IV
No. of sub-carriers	1536	384	192	768
Sub-carrier spacing	1kHz	4kHz	8kHz	2kHz
Symbol time	1.246ms	311.5us	155.8us	623us
Guard time	246us	61.5us	30.8us	123us
Carrier frequency	<375MHz	<1.5GHz	<3GHz	<1.5GHz
Transmitter separation	<96km	<24km	<12km	<48km

The DAB transmitted data consists of number of signals sampled at a rate of 48 kHz with a 22-bit resolution. This signal is then compressed at rates ranging from 32 to 384 kbps, depending upon the desired quality. The resulting digital data is then divided into frames of 24 ms. DAB uses differential QPSK modulation for the sub-carriers. A nullsymbol indicates the start of the frame. A reference OFDM symbol is then sent to serve as a starting point for the differential decoding of the QPSK subcarriers. Differential Modulation avoids the use of complicated phase-recovery schemes. DAB uses a rate quarter convolutional code with a constraint length of 7 for error-correction. Interleaving is used to separate the coded bits in the frequency domain as much as possible, which avoids large error bursts in the case of deep fades affecting a group of sub-carriers.

In DVB-H: Digital Broadcast Services to Handheld Devices

Digital Video Broadcasting (DVB) is a set of internationally accepted standards for digital television. DVB-H is one of the established mobile TV formats. It permits transmission of very large files and can operate on 5, 6, 7 or 8 MHz bandwidth. DVB-H uses OFDM air interface technology, and includes a technique for power reduction in the tuner. It uses time-slicing so that the tuner can be switched off most of the time and is only on during short transmission bursts. This allows the tuner to operate over a reduced input bandwidth and also conserves power. OFDM is a very good choice for a mobile TV air interface. It offers good spectral efficiency, immunity to multi-path, good mobile performance, and it works well in single-frequency networks such as those planned for mobile TV. The structure of DVB-H is depicted in fig. 2.

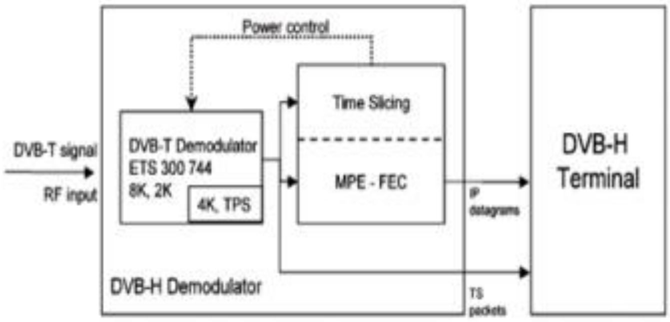

Figure 2: Conceptual structure of a DVB-H receiver [8]

It includes a DVB-H receiver (a DVB-T demodulator, a time-slicing module, and an optional MPE-FEC module) and a DVB-H terminal. The DVB-T demodulator recovers the MPEG-2 transport stream (TS) packets from the received DVB-T RF signal [8]. It offers three transmission modes: 8K, 4K, and 2K with the corresponding signalling. The time-slicing module controls the receiver to decode the wanted service and shut off during the other service bits. It aims to reduce receiver power consumption while also enabling a smooth and seamless frequency handover. The MPE-FEC module, provided by DVB-H, offers in addition to the error correction in the physical layer transmission, a complementary FEC function that allows the receiver to cope with particularly difficult reception situations. The advantages of DVB-H are as follows:

- Carriers - In DVB-H, carriers can use any additional spectrum that they might own for DVB-H broadcasting and be an infrastructure player.
- Spectrum Availability - In U.S., DVB-H will be organized using clear and "ready-for-use" spectrum available today, without interfering with existing analog TV stations or other TV or wireless services.

Advantages and disadvantages

Advantages of OFDM are listed as follows:

- OFDM makes resourceful utilization of the spectrum by overlapping. By dividing the channel into narrowband flat fading sub channels, OFDM is more resistant to frequency selective fading than single carrier systems.
- It can easily adapt to severe channel conditions without complex time-domain equalization.
- It reduces ISI and IFI through use of a cyclic prefix and fading caused by multipath

propagation.
• Using sufficient channel coding and interleaving lost symbols can be recovered.
• Channel equalization becomes simpler than by using adaptive equalization techniques with single carrier systems.
• OFDM is computationally capable by using FFT techniques to implement the modulation and demodulation functions.
• It is less sensitive to sample timing offsets than single carrier systems are.
• It is robust against narrow-band co-channel interference.
• Unlike conventional FDM, tuned sub-channel receiver filters are not required.
• It facilitates single frequency networks (SFNs); i.e., transmitter macro diversity.

The disadvantages are as follows:
• The OFDM signal has a noise like amplitude with a very large dynamic range; hence it requires RF power amplifiers with a high peak to average power ratio.
• It is more sensitive to carrier frequency offset and drift than single carrier systems are due to leakage of the DFT.
• It is sensitive to Doppler shift.
• It requires linear transmitter circuitry, which suffers from poor power efficiency.
• It suffers loss of efficiency caused by cyclic prefix.

Conclusion
OFDM has promising future in wireless networks and mobile communications. Growth in number of worldwide customers for wireless networks and ever-increasing demand for large bandwidth has given birth to this technology. OFDM is already playing an important role in WLAN and will be part of MAN too. In coming years, it will surely dominate the communication industry. Also, Wimax and 802.20 use OFDM-MIMO, which is emerging as the main technology for future cellular packet data networks, including 3GPP long-term evolution and 3GPP2 air interface evolution as well. Although OFDM has proven itself with packet-based data, it is not yet clear whether the technology can either handle large numbers of voice customers or work with voice and data as well as CDMA.

Massive MIMO Wireless Networks: Abstract: Massive multiple-input-multiple-output (MIMO) systems use few hundred antennas to simultaneously serve large number of wireless broadband terminals. It has been incorporated into standards like long term evolution (LTE) and IEEE802.11 (Wi-Fi). Basically, the more the antennas, the better shall be the performance. Massive MIMO systems envision accurate beamforming and decoding with simpler and possibly linear algorithms. However, efficient signal processing techniques have to be used at both ends to overcome the signalling overhead complexity. There are few fundamental issues about massive MIMO networks that need to be better understood before their successful deployment. In the third part of this lecture, we present a summarized review of massive MIMO homogeneous, and heterogeneous systems, highlighting key system components, pros, cons, and research directions.

Introduction

According to CISCO, an American multinational technology company, by 2020, more people (5.4 B) will have mobile phones than have electricity (5.3 B), running water (3.5 B) and cars (2.8 B). In addition, 75% of the mobile data traffic will be bandwidth-hungry video. Users will expect wireline quality in wireless services and higher bit rates and more reliable connections will be mandatory. While conventional techniques struggling to provide these bit rates, massive multiple-input-multiple-output (MIMO) systems promise 10 s of Gbps data rates to support real-time wireless multimedia services without occupying much additional spectrum. Massive MIMO technology has got much attraction lately as it promises truly broadband wireless networks. Massive MIMO systems use base station (BS) antenna arrays, with few hundred elements, simultaneously serving many tens of active terminals (users) using the same time and frequency resources.

Background

It is well known that, in classical MIMO, multiple antennas at both ends exploit wireless channel diversity to provide more reliable high-speed connections. Massive MIMO (also known as Large-Scale Antenna Systems, Very Large MIMO, Hyper MIMO, and Full-Dimension MIMO) makes a bold development from current practice using a very large number of service antennas (e.g., hundreds or thousands) that are operated fully coherently and adaptively. Figure 1 shows the speed improvement of wireless networks over the years starting from single-input-single-output (SISO) systems, single user (SU) and multiple users (MU) MIMO networks. MU-MIMO systems already provide significant advantages over earlier systems. Massive MIMO aims to further enhance this (to 10 Gbps and more) using hundreds of antennas exploiting advances in parallel digital signal processing and high-speed electronics. Extra antennas help with focusing the transmission and reception of signal energy into ever-smaller regions of space. This brings huge improvements in throughput and energy efficiency, in particular when combined with simultaneous scheduling of numerous user terminals (e.g., tens or hundreds).

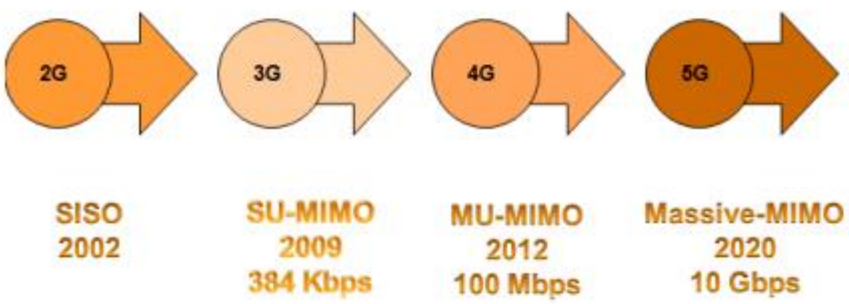

Figure 1. Evolving speed of wireless networks.

The more the BS antennas used, the more the data streams can be released to serve more terminals, reducing the radiated power, while boosting the data rate. This will also improve link reliability through spatial diversity and, provide more degrees of freedom in the spatial domain, and improve the performance irrespective of the noisiness of the measurements. In addition, because massive MIMO systems have a broad range of states of freedom, and greater selectivity in transmitting and receiving the data streams, interference cancellation is enhanced. BSs can relatively easily avert transmission into undesired directions to alleviate harmful interference which, leads to low latency as well. In addition, massive MIMO makes a proper use of beamforming techniques to reduce fading drops; this further boosts signal-to-noise-ratio (SNR), bit rate and reduces latency. Furthermore, increasing the number of BS antennas above the number of active users leads to higher throughput. Channel estimation quality per antenna also improves with the number of BS antennas especially in the presence of high correlation among the antennas which is very typical. In addition, the eigenvalue histogram of a single implementation converges to the average asymptotic eigenvalue distribution. This leads to the possibility of employing simple low complexity detection techniques while preserving an excellent performance. In addition, the channel becomes more predestined and random detectors matrices are readily solved.

Aggressive spatial multiplexing in massive MIMO systems leads to an impressive improvement in the network capacity by minimizing multiuser interference by steering the signal accurately in the right direction. Massive MIMO systems concentrate the released energy into small user centric zones, which dramatically increases the throughput and the energy efficiency. Since all of the users can take part in the multiplexing gain, costly antenna array deployments are only necessary on the BS side, which saves on costs by sharing. This also leaves the user equipment less complex, often with a single antenna. A higher number of BS antennas revokes the effects of uncorrelated noise and small-scale fading, and lowers the required transmitted energy per bit. The propagation medium minimally affects the performance of a massive MIMO system because of multi-user diversity. Due to the advantages and popularity of massive MIMO, recently there has been an increase in papers written on this area. *Networked MIMO and Massive MIMO* MIMO systems can be cooperative or non-cooperative. Cooperating systems are often called *Networked MIMO*, where a certain user is served by all BSs within its range of operation. The typical massive MIMO BSs do not cooperate in this sense. Both systems mitigate interferences of multi cellular wireless networks in separate ways and are not to be confused with each other. Networked MIMO emulates distributed antenna arrays by creating clusters of connected BSs. Note that each BS has a relatively small number of antennas only. Channel state information (CSI) as well as data are shared among the collaborating BSs through backhaul links. This contributes to interference cancellation, and then data is passed to the scheduled downlink users cooperatively from the BSs (sometimes using beamforming). In contrary, massive MIMO systems have substantial (M) number of antennas per BS, simultaneously serving a much smaller (K) number of users.

Massive MIMO in Wireless Sensor Networks

Wireless sensor networks (WSN) are special kinds of monitoring networks, aiming at detecting, measuring, monitoring certain physical phenomena, such as temperature, humidity, pressure, vibration, etc. Each device in the WSN is termed as a node that exchanges information with its neighbour. Typically, nodes have limited connectivity and energy resources. All data will be poured into a BS node, or a sink, which in turn relays the information to an outside user, or a server to process it. WSN nodes are small in size, cheap in cost, and do not employ complicated processing units, except the sink node. WSN may be composed of hundreds or thousands of nodes to provide coverage on a large-scale basis. Recently, several research efforts have been addressed to discuss the benefit of introducing a massive number of antennas at the BS, or the sink node. Multiple antennas at the BS improve the detection performance, the estimation performance, and energy efficiency, even when using simpler algorithms, and linear receivers with partial CSI knowledge.

In further studies, the detection, and estimation performances of a Gaussian signal communicating over a coherent multiple access channel in a WSN have a massive MIMO BS, or a fusion centre (FC). The Neyman–Pearson detectors and the linear minimum mean squared error (LMMSE) estimation detectors that require full CSI were also examined and Significant performance gains were achieved at low sensor transmit power levels. However, the energy detector shows improvement in gain under both low and high sensor power assumptions. In addition, in, the authors compared the performance of a low-cost energy detector to that of an expensive complex optimal detector, in a WSN having multiple antennas at the FC, both analytically and by simulation. Finally, the authors in optimized the transmission power at each node of a WSN having multiple antennas at the FC, using two different scenarios—in correlated and in uncorrelated fading channels with noise. The authors proved that the total power consumption at the nodes is saved as the number of antennas increases.

Channel Equalization Techniques for Wireless Communications Systems

Introduction and Motivation

In bandlimited, high data rate digital communication systems, equalizers are important devices. Their function is to restore the transmitted information, i.e., the information at the channel input, decreasing or eliminating channel interference. A large variety of techniques have been developed in the last 70 years, following the evolution of communication systems.

Initially, researchers were interested in guaranteeing the correct transmission of information between two points, leading to the so-called single-input/single-output (SISO) systems. The foundation of equalization and adaptive filtering was developed in this context. Considering that a communication channel can be modeled as a linear time-invariant (LTI) filter, whose output is added to a noise, the received signal is given by

$$x[n] = \sum_{k=-\infty}^{\infty} h[k]s[n-k] + v[n], \tag{8.1}$$

where $h[n]$ is the channel impulse response, $s[n]$ is the transmitted symbol, and $v[n]$ is the additive white Gaussian noise (AWGN). Rearranging terms to emphasize the presence of the symbol $s[n]$

$$x[n] = h[0]s[n] + \sum_{k=-\infty, k\neq 0}^{\infty} h[k]s[n-k] + v[n] \tag{8.2}$$

enables the observation that the received message is in fact given by the original signal added to noise and to a third term that is a function of delayed versions of the transmitted symbol. This term is the so-called intersymbol interference (ISI). One of the main tasks of an equalizer is to eliminate or at least to reduce its effect, and also that of the noise, so that the desired message can be recovered correctly. In fact, if the equalizer may be implemented as an LTI filter, then a perfect equalization is achieved when the following equation is satisfied:

$$y[n] = As[n-\Delta], \tag{8.3}$$

where $y[n]$ is the equalizer output, A is a gain, and Δ is a delay. Note that this solution would only be possible if the convolution between the channel and the equalizer impulse responses resulted in a vector of the form $[0 \ldots 0\ 1\ 0 \ldots 0]$, that is, a null vector except for the position where $n = \Delta$. For this reason, this solution is known as the zero-forcing (ZF) solution. Unfortunately, this solution is often impossible to be attained, specially due to the structures used to model the channel and the equalizer filters. This linear equalization process is exemplified in Fig. 8.1. For channels with deep spectral nulls, only the use of non-linear structures may lead to satisfactory equalization results.

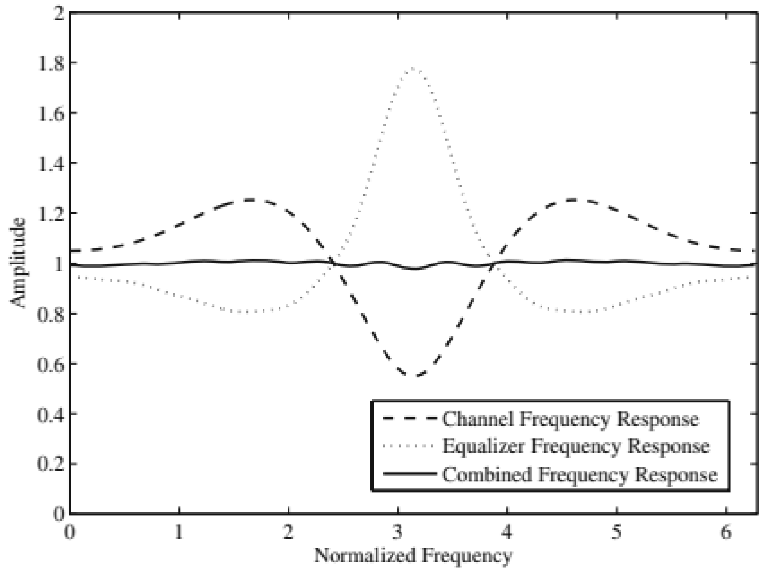

Fig. 8.1 Exemplifying the linear equalization of a channel.

When a wireless transmission is considered, the channel will not only introduce ISI but also something called fading, which results from the destructive interference between multiple paths. In such a context, it is important to take into account the user mobility, which causes a frequency offset due to the Doppler effect and that will cause phase and power fluctuations along the time. Equalizers must adapt to these channel variations. The exploitation of time diversity and/or frequency diversity becomes crucial for attaining good-quality higher data rate transmissions in lower signal-to-noise ratio (SNR). Soon enough, researchers found still another way of increasing quality: the exploitation of space diversity. Instead of transmitting through one antenna, why not using more than one? Or, similarly, if one antenna is used for transmission, why not use more than one to receive the information? This resulted in the so-called multiple-input single-output (MISO) and single-input multiple-output (SIMO) systems. New equalization techniques were proposed leading to important decreases in bit-error rate at the receiver output. Finally, generalizing the mentioned cases, we may consider several antenna for transmission and for reception, leading to the multiple-input multiple-output (MIMO) systems.

REFERENCE
1. Wireless Networking in the Developing World: A practical guide to planning and building low-cost telecommunications infrastructure (PDF) (2nd ed.). Hacker Friendly LLC. 2007. p. 425.

2. Pahlavan, Kaveh; Levesque, Allen H (1995). Wireless Information Networks. John Wiley & Sons. ISBN 0-471-10607-0.

3. Geier, Jim (2001). Wireless LANs. Sams. ISBN 0-672-32058-4.

4. Goldsmith, Andrea (2005). Wireless Communications. Cambridge University Press. ISBN 0-521-83716-2.

5. Molisch, Andreas (2005). Wireless Communications. Wiley-IEEE Press. ISBN 0-470-84888-X.

6. Pahlavan, Kaveh; Krishnamurthy, Prashant (2002). Principles of Wireless Networks – a Unified Approach. Prentice Hall. ISBN 0-13-093003-2.

7. Rappaport, Theodore (2002). Wireless Communications: Principles and Practice. Prentice Hall. ISBN 0-13-042232-0.

8. Rhoton, John (2001). The Wireless Internet Explained. Digital Press. ISBN 1-55558-257-5.

9. Tse, David; Viswanath, Pramod (2005). Fundamentals of Wireless Communication. Cambridge University Press. ISBN 0-521-84527-0.

10. Pahlavan, Kaveh; Krishnamurthy, Prashant (2009). Networking Fundamentals – Wide, Local and Personal Area Communications. Wiley. ISBN 978-0-470-99290-6.