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Abstract - Adaptive modulation and diversity combining represent very important adaptive solutions for the future generations of communication systems. In order to improve the performance and the efficiency of wireless communication systems these two techniques have been recently used jointly in new schemes named joint adaptive modulation and diversity combining .The highest spectral efficiency with the lowest possible combining complexity, given the fading channel conditions and the required error rate performance. Increase the spectral efficiency with a slight increase in the average number of combined path for the low signal to noise ratio (SNR) range while maintaining compliance with the bit error rate (BER).

***Keywords** - BER, OFDM, WCDMA, DAB, DVB.*

I. INTRODUCTION

Wireless communications is an emerging field, which has seen enormous growth in the last several years. The huge uptake rate of mobile phone technology, Wireless Local Area Networks (WLAN) and the exponential growth of the Internet have resulted in an increased demand for new methods of obtaining high capacity wireless networks. Most WLAN systems currently use the IEEE802.11b standard, which provides a maximum data rate of 11 Mbps. Newer WLAN standards such as IEEE802.11a and HiperLAN2, are based on OFDM technology and provide a much higher data rate of 54 Mbps. However systems of the near future will require WLANs with data rates of greater than 100 Mbps, and so there is a need to further improve the spectral efficiency and data capacity of OFDM systems in WLAN applications.

For cellular mobile applications, we will see in the near future a complete convergence of mobile phone technology, computing, Internet access, and potentially many multimedia applications such as video and high quality audio. In fact, some may argue that this convergence has already largely occurred, with the advent of being able to send and receive data using a notebook computer and a mobile phone.

Although this is possible with current 2G (2nd Generation) Mobile phones, the data rates provided are very low (9.6 kbps – 14.4 kbps) and the cost is high (typically $0.20 - $1.30 AUD per minute), limiting the usefulness of such a service.

The goal of third and fourth generation mobile networks is to provide users with a high data rate, and to provide a wider range of services, such as voice communications, videophones, and high speed Internet access. The higher data rate of future mobile networks will be achieved by increasing the amount of spectrum allocated to the service and by improvements in the spectral efficiency. OFDM is a potential candidate for the physical layer of fourth generation mobile systems. This thesis presents techniques for improving the spectral efficiency of OFDM systems applied in WLAN and mobile networks.

THIRD GENERATION WIRELESS SYSTEMS

Third generation mobile systems such as the Universal Mobile Telecommunications System (UMTS) [1], [2], [3], [4] and CDMA2000 [6] will be introduced over the next 1-5 years (2002 onwards) [5]. These systems are striving to provide higher data rates than current 2G systems such as the Global System for Mobile communications (GSM) and IS-95. Second generation systems are mainly targeted at providing voice services, while 3rd generation systems will shift to more data oriented services such as Internet access. Third generation systems use Wide-band Code Division Multiple Access (WCDMA) as the carrier modulation scheme [10]. This modulation scheme has a high multipath tolerance, flexible data rate, and allows a greater cellular spectral efficiency than 2G systems. Third generation systems will provide a significantly higher data rate (64 kbps – 2 Mbps) [1] than second-generation systems (9.6 – 14.4 kbps). The higher data rate of 3G systems will be able to support a wide range of applications including Internet access, voice communications and mobile videophones. In addition to this, a large number of new applications will emerge to utilise the permanent network connectivity, such as wireless appliances, notebooks with built in mobile phones, remote logging, wireless web cameras, car navigation systems, and so forth. In fact most of these applications will not be limited by the data rate provided by 3G systems, but by the cost of the service.

The demand for use of the radio spectrum is very high, with terrestrial mobile phone systems being just one of many applications vying for suitable bandwidth. These applications require the system to operate reliably in non-line-of-sight environments with a propagation distance of 0.5 - 30 km, and at velocities up to 100 km/hr or higher. This operating environment limits the maximum RF frequency to 5 GHz, as operating above this frequency results in excessive channel path loss, and excessive Doppler spread at high velocity. This limits the spectrum available for mobile applications, making the value of the radio spectrum extremely high. In Europe auctions of 3G licenses of the radio spectrum began in 1999. In the United Kingdom, 90 MHz of bandwidth [8] was auctioned off for £22.5 billion [9]. In Germany the result was similar, with 100 MHz of bandwidth raising $46 billion (US) [7]. This represents a value of around $450 Million (US) per MHz. The length of these license agreements is 20 years [8] and so to obtain a reasonable rate of return of 8% on investment, $105 Million (US) per MHz must be raised per year. It is therefore vitally important that the spectral efficiency of the communication system is maximised, as this is one of the main limitations to providing a low cost high data rate service.

4TH GENERATION SYSTEMS AND BEYOND

Research has just recently begun on the development of 4th generation (4G) mobile communication systems. The commercial rollout of these systems is likely to begin around 2008 - 2012, and will replace 3rd generation technology. Few of the aims of 4G networks have yet been published, however it is likely that they will be to extend the capabilities of 3G networks, allowing a greater range of applications, and improved universal access. Ultimately 4G networks should encompass broadband wireless services, such as High Definition Television (HDTV) (4 - 20 Mbps) and computer network applications (1 - 100 Mbps). This will allow 4G networks to replace many of the functions of WLAN systems. However, to cover this application, cost of service must be reduced significantly from 3G networks. The spectral efficiency of 3G networks is too low to support high data rate services at low cost. As a consequence one of the main focuses of 4G systems will be to significantly improve the spectral efficiency. In addition to high data rates, future systems must support a higher Quality Of Service (QOS) than current cellular systems, which are designed to achieve 90 - 95% coverage [11], i.e. network connection can be obtained over 90 - 95% of the area of the cell. This will become inadequate as more systems become dependent on wireless networking. As a result 4G systems are likely to require a QOS closer to 98 - 99.5%.

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

Orthogonal Frequency Division Multiplexing (OFDM) is an alternative wireless modulation technology to CDMA.

OFDM has the potential to surpass the capacity of CDMA systems and provide the wireless access method for 4G systems. OFDM is a modulation scheme that allows digital data to be efficiently and reliably transmitted over a radio channel, even in multipath environments. OFDM transmits data by using a large number of narrow bandwidth carriers. These carriers are regularly spaced in frequency, forming a block of spectrum. The frequency spacing and time synchronisation of the carriers is chosen in such a way that the carriers are orthogonal, meaning that they do not cause interference to each other. This is despite the carriers overlapping each other in the frequency domain. The name 'OFDM' is derived from the fact that the digital data is sent using many carriers, each of a different frequency (Frequency Division Multiplexing) and these carriers are orthogonal to each other, hence Orthogonal Frequency Division Multiplexing. The origins of OFDM development started in the late 1950's with the introduction of Frequency Division Multiplexing (FDM) for data communications. In 1966 Chang patented the structure of OFDM and published the concept of using orthogonal overlapping multi-tone signals for data communications. In 1971 Weinstein introduced the idea of using a Discrete Fourier Transform (DFT) for implementation of the generation and reception of OFDM signals, eliminating the requirement for banks of analog subcarrier oscillators. This presented an opportunity for an easy implementation of OFDM, especially with the use of Fast Fourier Transforms (FFT), which are an efficient implementation of the DFT. This suggested that the easiest implementation of OFDM is with the use of Digital Signal Processing (DSP), which can implement FFT algorithms. It is only recently that the advances in integrated circuit technology have made the implementation of OFDM cost effective. The reliance on DSP prevented the wide spread use of OFDM during the early development of OFDM. It wasn't until the late 1980's that work began on the development of OFDM for commercial use, with the introduction of the Digital Audio Broadcasting (DAB) system.

II. DIGITAL AUDIO BROADCASTING

DAB was the first commercial use of OFDM technology [19], [20]. Development of DAB started in 1987 and services began in U.K and Sweden in 1995. DAB is a replacement for FM audio broadcasting, by providing high quality digital audio and information services. OFDM was used for DAB due to its multipath tolerance. Broadcast systems operate with potentially very long transmission distances (20 -100 km). As a result, multipath is a major problem as it causes extensive ghosting of the transmission. This ghosting causes Inter-Symbol Interference (ISI), blurring the time domain signal. For single carrier transmissions the effects of ISI are normally mitigated using adaptive equalisation. This process uses adaptive filtering to approximate the impulse response of the radio channel. An inverse channel response filter is then used to recombine the blurred copies of the symbol bits. This process is however complex and slow due to the locking time of the adaptive equaliser. Additionally it becomes increasing difficult to equalise signals that suffer ISI of more than a couple of symbol periods. OFDM overcomes the effects of multipath by breaking the signal into many narrow bandwidth carriers. This results in a low symbol rate reducing the amount of ISI. In addition to this, a guard period is added to the start of each symbol, removing the effects of ISI for multipath signals delayed less than the guard period (see section 2.3 for more detail). The high tolerance to multipath makes OFDM more suited to high data transmissions in terrestrial environments than single carrier transmissions.

Parameter	Transmission Mode			
	I	II	III	IV
Bandwidth	1.536 MHz	1.536 MHz	1.536 MHz	1.536 MHz
Modulation	DQPSK	DQPSK	DQPSK	DQPSK
Frequency Range (Mobile reception)	≤ 375 MHz	≤ 1.5 GHz	≤ 3 GHz	≤ 1.5 GHz
Number of subcarriers	1536	384	192	768
Symbol Duration	1000 μs	250 μs	125 μs	500 μs
Guard Duration	246 μs	62 μs	31 μs	123 μs
Total Symbol Duration	1246 μs	312 μs	156 μs	623 μs
Maximum Transmitter Separation for SFN	96 km	24 km	12 km	48 km

Table 1-1, DAB Transmission parameters for each transmission mode

Table 1-1 shows the system parameters for DAB. DAB has four transmission modes.

The transmission frequency, receiver velocity and required multipath tolerance all determine the most suitable transmission mode to use. Doppler spread is caused by rapid changes in the channel response due to movement of the receiver through a multipath environment. It results in random frequency modulation of the OFDM subcarriers, leading to signal degradation. The amount of Doppler spread is proportional to the transmission frequency and the velocity of movement. The closer the subcarriers are spaced together, the more susceptible the OFDM signal is to Doppler spread, and so the different transmission modes in DAB allow trade off between the amount of multipath protection (length of the guard period) and the Doppler spread tolerance.

The high multipath tolerance of OFDM allows the use of a Single Frequency Network (SFN), which uses transmission repeaters to provide improved coverage, and spectral efficiency. For traditional FM broadcasting, neighbouring cities must use different RF frequencies even for the same radio station, to prevent multipath causes by rebroadcasting at the same frequency. However, with DAB it is possible for the same signal to be broadcast from every area requiring coverage, eliminating the need for different frequencies to be used in neighbouring areas**.**

The data throughput of DAB varies from 0.6 - 1.8 Mbps depending on the amount of Forward Error Correction (FEC) applied. This data payload allows multiple channels to be broadcast as part of the one transmission ensemble. The number of audio channels is variable depending on the quality of the audio and the amount of FEC used to protect the signal. For telephone quality audio (24 kbps) up to 64 audio channels can be provided, while for CD quality audio (256 kb/s), with maximum protection, three channels are available.

III. DIGITAL VIDEO BROADCASTING

The development of the Digital Video Broadcasting (DVB) standards was started in 1993 [14]. DVB is a transmission scheme based on the MPEG-2 standard, as a method for point to multipoint delivery of high quality compressed digital audio and video. It is an enhanced replacement of the analogue television broadcast standard, as DVB provides a flexible transmission medium for delivery of video, audio and data services [17]. The DVB standards specify the delivery mechanism for a wide range of applications, including satellite TV (DVB-S), cable systems (DVB-C) and terrestrial transmissions (DVB-T) [15]. The physical layer of each of these standards is optimised for the transmission channel being used. Satellite broadcasts use a single carrier transmission, with QPSK modulation, which is optimised for this application as a single carrier allows for large Doppler shifts, and QPSK allows for maximum energy efficiency [16]. This transmission method is however unsuitable for terrestrial transmissions as multipath severely degrades the performance of high-speed single carrier transmissions. For this reason, OFDM was used for the terrestrial transmission standard for DVB. The physical layer of the DVB-T transmission is similar to DAB, in that the OFDM transmission uses a large number of subcarriers to mitigate the effects of multipath. DVB-T allows for two transmission modes depending on the number of subcarriers used [18]. Table 1-2 shows the basic transmission parameters for these two modes. The major difference between DAB and DVB-T is the larger bandwidth used and the use of higher modulation schemes to achieve a higher data throughput. The DVB-T allows for three subcarrier modulation schemes: QPSK, 16-QAM (Quadrature Amplitude Modulation) and 64- QAM; and a range of guard period lengths and coding rates. This allows the robustness of the transmission link to be traded at the expense of link capacity. Table 1-3 shows the data throughput and required SNR for some of the transmission combinations. Lowered to meet DVB-T is a uni-directional link due to its broadcast nature. Thus any choice in data rate verses robustness affects all receivers. If the system goal is to achieve high reliability, the data rate must be the conditions of the worst receiver. This effect limits the usefulness of the flexible nature of the standard. However if these same principles of a flexible transmission rate are used in bi-directional communications, the data rate can be maximised based on the current radio conditions**.** Additionally for multiuser applications, it can be optimised for individual remote transceivers

Parameter	2k Mode	8k Mode
Number subcarriers	1705	6817
Useful Symbol Duration (T_u)	896 µs	224 µs
Carrier Spacing ($1/T_u$)	1116 Hz	4464 Hz
Bandwidth	7.61 MHz	7.61 MHz

Table 1-2, DVB transmission parameters

Subcarrier Modulation	Code Rate	SNR for BER = 2×10^{-4} after Viterbi (dB)		Bit rate (Mbps) Guard Period (Fraction of Useful symbol duration)	
		Gaussian Channel	Rayleigh Channel	1/4	1/32
QPSK	½	3.1	5.4	4.98	6.03
QPSK	7/8	7.7	16.3	8.71	10.56
16-QAM	½	8.8	11.2	9.95	12.06
16-QAM	7/8	13.9	22.8	17.42	21.11
64-QAM	½	14.4	16.0	14.93	18.10
64-QAM	7/8	20.1	27.9	26.13	31.67

Table 1-3, SNR required and net bit rate for a selection of the coding and modulation combinations for DVB

IV. MULTIUSER OFDM

DAB and DVB systems are only uni-directional from the base station to the users. Not much work has been done on using OFDM for two-way communications or for multiuser applications. These applications include wireless modems, Wireless Local Area Networks (WLAN's), Wireless Local Loop (WLL), mobile phones, and mobile high speed internet. This thesis aims to look at applying OFDM to such applications, and to look at the resulting advantages and problems. This thesis also presents some new techniques that can be used to improve broadcast and multiuser OFDM systems. The performance of adaptive modulation and adaptive user allocation schemes in a multiuser OFDM system. These techniques improve the spectral efficiency and QOS. Fattouche patented a method for implementing a wireless multiuser OFDM system in 1992, predating any published research in this field. This system used half duplex Time Division Multiplexing (TDM) to allow multiuser access, with the base stations and portable units taking turns to transmit. Carrier modulation was fixed and used D8-PSK (Differential 8 Phase Shift Keying). The system was bandwidth limited by using a raised cosine guard period. Fattouche is the founder WiLan Inc., which is one of the few companies currently producing multiuser OFDM modems. Williams and Prodan [82], patented the use of multiuser OFDM in cable applications in 1995. This introduced the use of a hybrid user allocation, using Frequency Division Multiplexing (FDM) and TDM. In this system the users were allocated time and frequency slots depending on the data demand. This patent however, fails to address problem of obtaining and maintaining accurate time and frequency synchronisation between users, which is critical for maintaining orthogonality between users. Cimini, Chuang, Sollenberger outlined an Advanced Cellular Internet Service using multiuser OFDM. The aim of this system was to provide Internet access at a data rate of 1 – 2 Mbps. This system uses time synchronised base stations, which are allocated time slots in a self-organising fashion. These base station time slots are then broken down in to time slots for users. In addition to TDM, users are allocated subcarriers dynamically based on the channel Signal to Interference Ratio (SIR), to allow minimisation of inter-cellular interference.

Wahlqvist described one possible implementation of multiuser OFDM in a wireless environment, outlining a user allocation scheme where users were allocated small blocks of time and frequency. In this scheme, each transmission block consists of a small group of subcarriers, (5 - 10) and a small number of symbols, about 11 in length. The aim of this structure is to allocate time and frequency slots to utilise the high correlation between neighbouring subcarriers, and the small channel variation between a small group of symbols. This allows the block to be characterised with a simple pilot tone structure.

V. CONCLUSION

The detail knowledge of a current key issue in the field of communications named Orthogonal Frequency Division Multiplexing (OFDM). We elaborated on the performance theory of the codes. First I developed an OFDM system model then try to improve the performance by applying forward error correcting codes to our uncoded system. From the study of the system, it can be concluded that we are able to improve the performance of uncoded OFDM by convolutional coding scheme.

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