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RECONFIGURABLE OFDM SYSTEMS

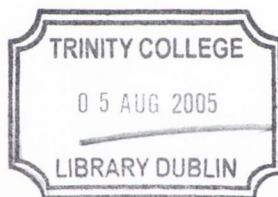
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A thesis submitted for the degree of

Doctor of Philosophy

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May 2005

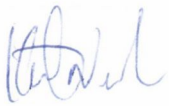


THESIS

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SUMMARY

Orthogonal Frequency Division Multiplexing (OFDM) is a multi-carrier wireless transmission technique. OFDM is used for robust, high quality and high data-rate music, voice, images, video, news and data broadcasts. It is also used for high data-rate Wireless Local Area Network (WLAN) communications links, and is expected to play an important part in emerging and future wireless communications schemes and devices. OFDM uses many carrier frequencies in unison, each transmitting a section of an input data sequence. This operation enables OFDM to achieve its high-data rate capabilities. The carrier frequencies may be spaced as close together as is theoretically possible, yet they do not interfere with one another. These carrier frequencies are known as orthogonal carriers. OFDM is more robust than single-carrier transmission systems in a multi-path fading channel environment due to its frequency-diversity and transmission structure characteristics.

This thesis provides an introduction to OFDM and shows that OFDM can be significantly enhanced using reconfigurable radio. The signal-processing core of the reconfigurable radio system presented in this thesis is implemented in software. The improved OFDM techniques enabled by the use of reconfigurable radio comprise a low-complexity OFDM synchronisation scheme, a novel frequency-hopping version of OFDM, a novel method of choosing which carrier-frequencies to use for transmission and a novel method of estimating the modulation scheme used for a received signal without *a priori* knowledge of the signal-type. Results and analysis are provided in order to demonstrate the capabilities and potential of these novel schemes and techniques.

The following publications directly relate to this thesis:

- [Nolan2001] Nolan, K., Doyle, L., O'Mahony, D. & Mackenzie, P. "Modulation Scheme Recognition Techniques for Software Radio on a General Purpose Processor Platform", in Proceedings of the First Joint IEI/IEE Symposium on Telecommunications Systems Research, Dublin, November 27th, 2001, 5pp
- [Nolan2002a] Nolan K.E, Doyle L., Mackenzie P., Ammann M.J., "Software Radio Modulation Scheme Recognition Techniques for ISI Channels", in Proceedings of the ISSC Irish Signals and

Systems Conference 2002, June 25-26, 2002, University College Cork and Cork Institute of Technology, Cork, Rep. Of Ireland, pp 263-268

- [Nolan2002b] Nolan,K.E., Doyle,L., Mackenzie,P. and O'Mahony,D., "Modulation Scheme Classification for 4G Software Radio Wireless Networks", in Proceedings of the IASTED International Conference on Signal Processing, Pattern Recognition, and Applications (SPPRA 2002), June 25-28, 2002, Crete, Greece, pp 25-31
- [Nolan2002c] Nolan,K.E., Doyle,L., O'Mahony,D. and Mackenzie, P., "Signal Space-based Adaptive Modulation for Software Radio", in Proceedings of the IEEE Wireless Communications and Networking Conference WCNC 2002, March 17-21 2002, Orlando, Florida, pp 510-515.
- [Nolan2003a] Nolan, K.; Mackenzie, P.; Doyle, L.; Flood, D., "Flexible Architecture Software Radio OFDM Transceiver System and Frame Synchronization Analysis", in Proceedings of the Global Telecommunications Conference, 2003. IEEE GLOBECOM '03, Vol.: 1, 1-5 Dec. 2003, pp 332 – 336
- [Nolan2003b] Nolan K.E, Mackenzie P., Doyle L. O'Mahony D., "OFDM/Flash-OFDM Reconfigurable Transceivers Using General Purpose Processors", in Proceedings of IEE DSP Enabled Radio Colloquium, Scotland, UK, 22nd-23rd September 2003. pp 407 – 418
- [Nolan2003c] Nolan K.E., Mackenzie P., Doyle L., Flood D., "Implementation of a General Purpose Processor Platform Based OFDM Software Radio System", in Proceedings of the Irish Signals and Systems Conference (ISSC) 2003, University Of Limerick, Rep. Of Ireland, July 1st -July 2nd 2003, pp 406 – 410
- [Nolan2003d] Nolan, K. E., Mackenzie P., Doyle L, Flood D., "A Pattern Recognition Approach for OFDM Frame Synchronization Using General Purpose Processors", in Proceedings of the IASTED International Conference on Signal Processing, Pattern Recognition and Applications (SPPRA 2003), June 30 -July 02 2003, Rhodes, Greece, pp 153 – 157
- [Nolan2004] Nolan, K.E, Mackenzie, P. & Doyle, L.E, "OFDM Sub-Carrier Allocation Algorithm For A Multiple User Data Enhanced Radio Server (MUDERS) Using A General Purpose Processor Platform", in Proceedings of the 4th Software Defined Radio (SDR) Forum Technical Conference, Phoenix, Arizona, Nov 15-18th, 2004.

ACKNOWLEDGEMENTS

I am deeply grateful for the guidance, insight, encouragement and patience of my supervisor, Linda Doyle, whose infectious enthusiasm and encouragement fuelled a very enjoyable time in Trinity College.

I would like to thank Prof. Donal O'Mahony for his inspirational comments and all of the members of the NTRG. I also wish to thank the Trinity Trust foundation who provided financial support for my paper presentations at the 2004 Software Defined Radio Forum technical conference. I also wish to thank IRCSET for financial support for the last two years of my research work and Enterprise Ireland for financial support during the first two years.

Many thanks to all of the staff and students in the Dept. of Electronic and Electrical Engineering, TCD, for creating such a friendly place to work. I also wish to thank Anil Kokaram for his helpful suggestions regarding signal-processing.

I wish to thank Philip, for his limitless knowledge of software, radio and music. A special thanks to Paul, Declan and especially to Tim for his words of wisdom and help.

I thank all of my family and friends. In particular, I wish to thank Enda, Jonathan, Colin, Lucy, Mick, Gavan and Colm for all their support.

I wish to dedicate this thesis to my parents Charles and Frances and my brother Derek who have provided endless encouragement and support.

Finally, my deepest thanks go to Deirdre for being a constant source of support, understanding and encouragement.

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ACRONYMS

AAC	Advanced Audio Coding	FAC	Fast Access Channel
ADC	Analogue to Digital Converter	FFT	Fast Fourier Transform
AM	Amplitude Modulation	FIB	Fast Information Block
ANN	Artificial Neural Network	FIC	Fast Information Channel
API	Application Programming Interface	FM	Frequency Modulation
ASIC	Application Specific Integrated Circuit	FPGA	Field Programmable Gate Array
ASK	Amplitude Shift Keying	FSK	Frequency Shift Keying
ATM	Asynchronous Transfer Mode	GPP	General Purpose Processor
AWGN	Additive White Gaussian Noise	GPS	Global Positioning System
BER	Bit Error Rate	GSM	Global System for Mobile communications
BPSK	Binary Phase Shift Keying	GUI	Graphical User Interface
CAZAC	Constant Amplitude Zero AutoCorrelation	HF	High Frequency
CELP	Code Excited Linear Prediction	ICI	Inter-Carrier Interference
CFO	Carrier Frequency Offset	ICM	Inter-Carrier Modulation
CIF	Common Interleaved Frame	IC	Integrated Circuit
CNR	Channel Noise Ratio	IF	Intermediate Frequency
COFDM	Coded Orthogonal Frequency Division Multiplexing	IFFT	Inverse Fast Fourier Transform
DAB	Digital Audio Broadcasting	IP	Internet Protocol
DAC	Digital to Analogue Converter	I	In-Phase
DC	Direct Current	Q	Quadrature
DECT	Digital Enhanced Cordless Telephone	IRIS	Implementing Radio in Software
DFT	Discrete Fourier Transform	ISI	Intersymbol Interference
DLL	Dynamic Link Library	ISM	Industrial, Scientific and Medical
DOFDM	Dynamic Orthogonal Frequency Division Multiplexing	LAN	Local Area Network
DPSK	Differential Phase Shift Keying	SMC	Spectrum Management Controller
DRM	Digital Radio Mondiale	SNR	Signal to Noise Ratio

DSP	Digital Signal Processor or Digital Signal Processing	SONET	Synchronous Optical Network
DVB-T	Digital Video Broadcasting – Terrestrial	S/P	Serial to Parallel
ETS	Enhanced Training Symbol	STS	Synchronous Transport Signal
E2R	End to End Reconfigurability	T-DAB	Terrestrial Digital Audio Broadcasting
MUF	Maximum Useable Frequency	MAC	Medium Access Control
NCO	Numerically Controlled Oscillator	MIMO	Multiple Input Multiple Output
OFDM	Orthogonal Frequency Division Multiplexing	ML	Maximum Likelihood
OOD	Object Oriented Design	MMI	Man Machine Interface
OS	Operating System	MSC	Main Service Channel
OTA	Over The Air	UHF	Ultra-High Frequency
PA	Power Amplifier	VHDL	Very High Speed Integrated Circuits Hardware
PAPR	Peak to Average Power Ratio	VHF	Very High Frequency
PHY	Physical Layer	WARC	World Administrative Radio Conference
PLCP	Physical Layer Convergence Function	WLAN	Wireless Local Area Network
PMD	PHY Medium Dependent	XML	eXtensible Markup Language
PN	Pseudo-random		
PPDU	PLCP Protocol Data Unit		
PSK	Phase Shift Keying		
QAM	Quadrature Amplitude Modulation		
QPSK	Quadrature Phase Shift Keying		
RAM	Random Access Memory		
RF	Radio Frequency		
RFCD	Reduced Form Constellation Diagram		
RISC	Reduced Instruction Set Computer		
RNC	Radio Network Controller		
SBR	Spectral Band Replication		
SDR	Software Defined Radio		

1

INTRODUCTION

1.1 Overview

Orthogonal Frequency Division Multiplexing (OFDM) systems may be enhanced using reconfigurable radio as an enabling technology for wireless channel adaptation, enhanced air-interface performance and improved receiver synchronisation techniques.

This thesis explains what OFDM is, before describing a radio development and implementation platform that forms the foundation for the enhanced-OFDM system. Finally, the novel techniques that enhance the performance and capabilities of OFDM using the chosen radio platform are presented.

1.1.1 Orthogonal Frequency Division Multiplexing (OFDM)

OFDM is a multi-carrier transmission scheme that enables information to be transmitted at high data-rates with a higher robustness to the effects of noise and fading than single-carrier transmission systems. OFDM is currently used for high data-rate applications including high-speed wireless networking applications and high quality digital audio and video broadcasting services. OFDM is a frequency-diversity-based transmission scheme and uses closely spaced orthogonal carriers that do not interfere with each other. A high data-rate input information stream is converted to a number of lower data-rate segments of the input stream. These lower-rate segments are simultaneously transmitted in parallel using the orthogonal carriers. An OFDM waveform is created in the frequency-domain initially before being converted to a real-valued time domain waveform using the Inverse Fast Fourier Transform (IFFT). This time-domain signal is a multiplex of all the orthogonal carriers, which may be de-multiplexed using the Fast Fourier Transform (FFT).

Chapter 1 explains how information can be represented using an OFDM signal and subsequently recovered.

An OFDM transceiver is difficult and complex to create as a purely-hardware device. Many unique and highly-accurate RF oscillators, filters and mixers are required, which result in an expensive and physically large radio. OFDM is ideally suited for implementation as a digital process however. The FFT operation may be used instead of separate RF oscillators and filters. This work presented in this thesis exploits this fact to enhance the capabilities of OFDM. Digital processes can be modified and controlled much more easily than comparable hardware versions. In order to alter the operation of a hardware device, physical modifications or even a complete replacement of the device is required. This is time-consuming and may be an expensive process. Chapter 4 outlines how radio design has evolved from pure hardware design with little or no reconfigurability options to high-reconfigurable software-based radio implementations.

In order to create a better OFDM system, one approach is to take account of the time-varying characteristics of a wireless communications channel and enable the radio to adapt according to these changing characteristics. A second method of improving the data throughput rate of OFDM is to reduce the amount of transmissions required to synchronise the remotely-located destination receivers. This thesis distinguishes between non-data carrying, synchronisation and control transmissions, known as transmission-overheads and data-bearing transmissions. The latter is the modulated information from the source, which may be voice, music, images, video, news or other service information. Non data-bearing signals are required for receiver synchronisation and to aid the receiver in the interpretation of the intercepted OFDM signals. As the transmission capacity of a wireless channel is limited, by reducing the amount of synchronisation-related transmissions, the amount of data that can be transmitted to the destination transceiver can be increased. Novel means of improving OFDM systems and reducing the transmission overheads of this system are presented in Chapters 7, 8, 9 and 10.

1.1.2 Reconfigurable Radio

A software radio is a software version of a typical hardware radio but in this case, many modes of transmission and reception may be implemented without affecting the physical size of the radio [Mitola1992]. This is achieved by re-programming the software radio in

order to implement the required mode of operation. A reconfigurable radio is a software radio but the structure and parameters associated with each stage of the software radio signal-processing chain may be dynamically modified [Pereira1999][Pereira2000]. A reconfigurable radio views the software radio as a sequence of signal-processing stages. The characteristics and objectives of each of these stages can be modified in response to a change in the wireless communications channel environment or the activity of the radio user.

This thesis presents a reconfigurable OFDM radio platform that is used to implement the OFDM modulation and demodulation processes in software. One of the main features of this reconfigurable radio platform is that it enables the software version of the OFDM radio to be dynamically modified. Dynamic modification of a radio allows the characteristics and objectives of a radio to be tailored to better suit a wireless environment. Radio adaptation to the wireless communications channel environment is one of the main focuses of the work presented in this thesis. The second main objective is to present new methods of achieving synchronisation and an interpretation of the received OFDM signal using reduced transmission overheads.

1.2 Summary of Contributions

As stated at the beginning of Section 1.1, the work presented in this thesis shows that OFDM systems may be enhanced using reconfigurable radio as an enabling technology for wireless channel adaptation and receiver synchronisation techniques. This thesis is proven through the following contributions, which may be broken down into three main parts:

- A. Introduction to OFDM and related products and applications.
- B. Overview of the chosen reconfigurable radio platform and OFDM implementation.
- C. Novel OFDM-enhancing techniques using this reconfigurable radio platform.

1.2.1 Part A

1.2.1.1 A Concise Introduction to OFDM

The first part of this thesis describes the main events and key research related to OFDM technology. A comprehensive explanation of the processes involved in the creation of an OFDM waveform and demodulation of a received baseband OFDM signal is presented.

An overview of the main commercially-available OFDM-based products and services is also given.

1.2.2 Part B

1.2.2.1 Reconfigurable OFDM Transceiver

The second part of this thesis presents the reconfigurable signal-processing platform, which forms the foundation for development and implementation of an enhanced reconfigurable OFDM system. An implementation of a reconfigurable OFDM system using a General Purpose Processor is also presented in order to create a strong foundation for the following work in this thesis. The structure and individual stages of this reconfigurable OFDM system can be dynamically re-arranged, replaced and modified to form a reconfigurable radio capable of continuously adapting to the wireless communications channel environment. This reconfigurable radio platform is the key enabling-technology allowing the realisation of the proposed enhanced OFDM-based communications technique proposals.

1.2.3 Part C

The third part of this thesis presents explanations and analysis of the novel techniques developed to enhance OFDM using the reconfigurable radio platform.

1.2.3.1 Frame Synchronisation Technique

OFDM signals are generally transmitted in groups of OFDM symbols called 'frames'. A received group of OFDM symbols contained in an OFDM frame can only be extracted and demodulated correctly if the beginning of each OFDM frame can be estimated accurately. As a result, frame synchronisation is a key part of the OFDM demodulation process. A low-complexity frame synchronisation technique, designed for the OFDM reconfigurable radio is presented in this thesis. This technique reduces the transmission-overhead required for frame synchronisation at the receiver and can also correct any carrier frequency offsets that may exist between the transmitter and receiver.

1.2.3.2 Dynamic OFDM technique

A novel frequency-hopping OFDM technique called Dynamic OFDM (DOFDM) is also presented. This technique enables the OFDM sub-carriers to be randomly distributed over

a wider frequency range. It will be shown that this scheme reduces the peak power of an OFDM transmission thus decreasing the possibility of non-linear Power Amplifier operation and undesired spurious radiation. It will also be described how this technique improves the security of transmitted information, increases the robustness of OFDM even further than standard OFDM in frequency-selective interference and noisy channel environments, and dramatically reduces the transmission overheads associated with this system.

1.2.3.3 Sub-carrier Allocation Technique

A traditional fixed-architecture OFDM transceiver uses pre-determined sub-carrier frequencies regardless of the interference and noise that may be present on one or more of these frequencies. A novel sub-carrier allocation technique is presented, which automatically avoids using sub-carriers with excessive noise and/or interference. This technique increases the robustness of OFDM by reducing the possibility of erroneous received data due to noise and/or interference. The second main contribution of this proposed sub-carrier allocation technique is a novel method of conveying details of the how the sub-carriers have been allocated, over a wireless link, to the remotely-located receivers.

1.2.3.4 Modulation Scheme Recognition Technique

The next major contribution in this thesis is a novel low-complexity modulation scheme recognition technique designed for reconfigurable radio implementation. This is a phase and statistical moment based classifier with an objective of presenting a low processing-time overhead to the reconfigurable radio receiver process.

1.3 Thesis Overview

Chapter 2 introduces OFDM and guides the reader through the key stages of the modulation and demodulation processes in a step by step fashion. The main OFDM-related products and applications are discussed in Chapter 3. Chapter 4 shows that radio design is moving from all-hardware design principles to software-oriented solutions. Each stage of this process is described and ordered in terms of increasing reconfigurability possibilities. A radio design and implementation platform that affords the highest degree of

radio reconfiguration for the exploration and development of an enhanced OFDM system is described in Chapter 5. This reconfigurable radio platform is called Implementing Radio In Software (IRIS) and operates using a General Purpose Processor (GPP) [Mackenzie2004]. Using this platform, an explanation of how an OFDM reconfigurable radio may be created is presented in Chapter 6. Chapter 7 is the first chapter relating to how OFDM may be improved using this reconfigurable radio platform. This chapter explains the OFDM frame synchronisation technique designed for implementation using the IRIS platform. Chapter 8 presents an explanation and analysis of the novel frequency-hopping OFDM technique called Dynamic OFDM (DOFDM). In Chapter 9, the novel sub-carrier allocation technique and novel low-complexity and low-transmission bandwidth receiver notification technique is presented. Chapter 10 shows how a reconfigurable receiver enables the implementation of the novel automatic modulation scheme classification technique. This technique enables the reconfigurable OFDM receiver to estimate the type of modulation used for each of the received OFDM sub-carriers without *a priori* knowledge of the modulation type. Chapter 11 summarises conclusions from this thesis and suggests how this work may be developed further.

2

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

2.1 Introduction

This thesis focuses on a multi-carrier modulation technique known as Orthogonal Frequency Division Multiplexing (OFDM), which enables robust high data-rate communications over time-varying wireless communication channels. The purpose of this chapter is to provide a technical description of an OFDM system that serves as a basis for the remainder of the thesis. The chapter begins by focussing on the background of OFDM, from its conception as a means of using more than one carrier frequency at a time to transmit information, to the current use of OFDM for high quality audio and video broadcasting and transceiver applications. Section 2.2 provides an overview of OFDM and Section 2.3 describes some of the main contributions towards the development of OFDM over the past four decades. Section 2.4 begins the exploration of the processes required to generate OFDM by examining the core processes required to create a typical OFDM transmitter. In Section 2.5, a brief description of how a generated OFDM signal may be transmitted on the band of interest is given. Following on from this, Section 2.6 describes the processing stages involved in the demodulation of an intercepted OFDM signal. In addition, a description of the challenges involved in synchronising the receiver to the transmitter, as well as outlining how OFDM offers robust communications over time-variant and noisy communication channels is given. A description of how OFDM may be used to improve the quality of communications in the frequency bands below 30 MHz is presented in Section 2.7. Section 2.8 concludes this chapter.

2.2 Overview of OFDM

Multi-carrier transmission uses two or more modulated signals, each carrying a single data stream over a communications channel. The multiplexed signals are independently demodulated in the receiver and then de-multiplexed resulting in the received bit stream.

OFDM is a type of frequency division multiplexing transmission scheme that allows users to transmit information across a communications channel at high data rates and offers a more robust alternative to traditional signal carrier-frequency transmission systems in noisy communication channels that suffer from fading. OFDM offers much more efficient use of the available Radio Frequency (RF) spectrum than single carrier-frequency communication systems due to the simultaneous use of multiple frequencies for data transmission. The channel equalisation processes required for OFDM are also less complex than for single-carrier transmission systems. Guard intervals are required during the OFDM transmission process to preserve carrier frequency orthogonality, which reduce the potential maximal spectral efficiency of this scheme, however [Wu1995].

The rapid increase in popularity for this scheme is due to the fact that it allows the available channel bandwidth to be used very efficiently. OFDM also enables very high data-rate transmission to be achieved. Multi-carrier OFDM systems also offer a greater degree of robustness against communications channel perturbations such as frequency-selective fading and multi-path fading effects, which are often prevalent with both fixed and mobile wireless transceivers. If the transceiver is moving, either in a car or being carried by a person, fading effects may degrade the received signal. OFDM offers extra protection against fading effects using the frequency-diversity properties of OFDM and methods used to share the transmission data over multiple frequencies.

In this thesis, the resulting multiplexed time-domain OFDM waveform is known as an OFDM symbol. An OFDM waveform is created in the frequency domain initially and then transformed into a time-domain waveform. This enables a far greater ability to control the characteristics of OFDM waveforms and the amount of frequency spectrum that is occupied by the signal than for traditional waveforms such as Amplitude Modulation (AM) and Frequency Modulation (FM). Examination of the frequency-spectrum of an OFDM symbol shows that the carriers are densely packed, in fact they can be as close to one another as theoretically possible, and the spectrum 'tails' of each carrier overlaps the rest of the carriers. The centre frequency of each carrier coincides with a null in the joint spectrum of all the other carriers, thus no inter-carrier interference is experienced from the adjacent carriers. Transmission of multiple signals using orthogonal frequencies means that there is no interference between the carriers although the individual sources may be transmitting simultaneously. OFDM offers highly robust communications over channels affected by impulse noise, multi-path fading thanks to the frequency diversity and also the

data block-interleaving technique used. Before describing these latter techniques and each step of the OFDM generation and demodulation process, it is valuable to describe how OFDM was developed and how it has evolved.

2.3 Background

Multi-carrier modulation began in the late 1950's with the implementation of the first voice band modem by Kineplex Collins. This modem could operate at 3000 bps, which was more than ten times the rate of other modems at the time, and indeed, for the following decade [Mosier1958]. In 1966, Chang and Saltzberg [OFDMPatent] from Bell Labs produced a paper and patent regarding OFDM. The main research work at this stage concerned unwieldy analogue OFDM schemes requiring a separate Radio Frequency (RF) oscillator and filter per carrier, and Weinstein and Ebert [Weinstein1971] proposed the use of the Fast Fourier Transform (FFT) and guard intervals for OFDM in 1971. This breakthrough meant that OFDM transceivers did not require the large amount of complex hardware required for analogue OFDM generation and demodulation. With the advances in algorithm design and implementation, OFDM transmission over telephone lines was proposed in 1980 by Peled and Ruiz [Peled1980]. As digital and computing techniques continued to evolve resulting in more efficient FFT processing units with reduced complexity, Leonard Cimini [Cimini1985] proposed the use of OFDM for mobile communications in 1985. OFDM was proposed for digital broadcasting schemes in 1985 by Alard and Lasalle [Alard1987] and in 1989, a complete OFDM broadcasting system was described by Rault, Castellain, and Le Floch [Rault1989]. The system proposed as a robust means of broadcasting sound to mobile devices affected by hostile communication channel conditions. This solution involved the use of OFDM in conjunction with convolution codes and Viterbi decoding. This method of transmission was a forerunner to the core transmission technology as is used in the digital audio broadcasting standards. The hardware implementation of this radio system was designed for the Ultra-High Frequency (UHF) band and in their paper presented at the Global Telecommunications Conference in 1989, it was reported that a field demonstration of digital audio broadcasting had been carried out at the World Administrative Radio Conference (WARC) on a Geo-stationary ORBit use (ORB) conference in 1998. This implementation was reported to be capable of processing sixteen 288 kbits/s stereo programmes using a bandwidth of 7 MHz. In 1990, Bingham [Bingham1990] examined the performance of OFDM over a wireless channel unaffected by distortion, in terms of the highest attainable bit rate for a specific SNR and

BER. An additional study on how the performance may be improved through the use of channel coding techniques was also presented.

In 1995, the European Telecommunications Standard Institute (ETSI) [ETSI1997a] [ETSI1997b] established the first OFDM-based standard (EN 301 401), and in 1997, published the finalised standards for DAB and Digital Video Broadcasting-Terrestrial (DVB-T) which are also OFDM-based schemes. In recent times, the appeal of OFDM has lead to its adoption in the North American IEEE802.11a standard, the IEEE802.11g standard and the European HIPERLAN/2 wireless Local Area Network (WLAN) standards in preference to Frequency Hopping Spread Spectrum (FHSS) [IEEE802.11a][IEEE802.11g][HIPERLAN/2]. Further explanations of these technologies are available later in this chapter.

2.4 OFDM Modulator

There are many ways to create an OFDM transceiver. The block diagram of one example of a typical OFDM transceiver is shown in Figure 2.1. This block diagram represents the OFDM transceiver model that will be used throughout this thesis. In this section, each stage of the OFDM generation and demodulation process is described. Initial explanations of the operation of an OFDM transceiver will begin by focussing on the transmitter, as highlighted in Figure 2.2, followed by a description of the operation of the OFDM receiver, as shown in Figure 2.18.

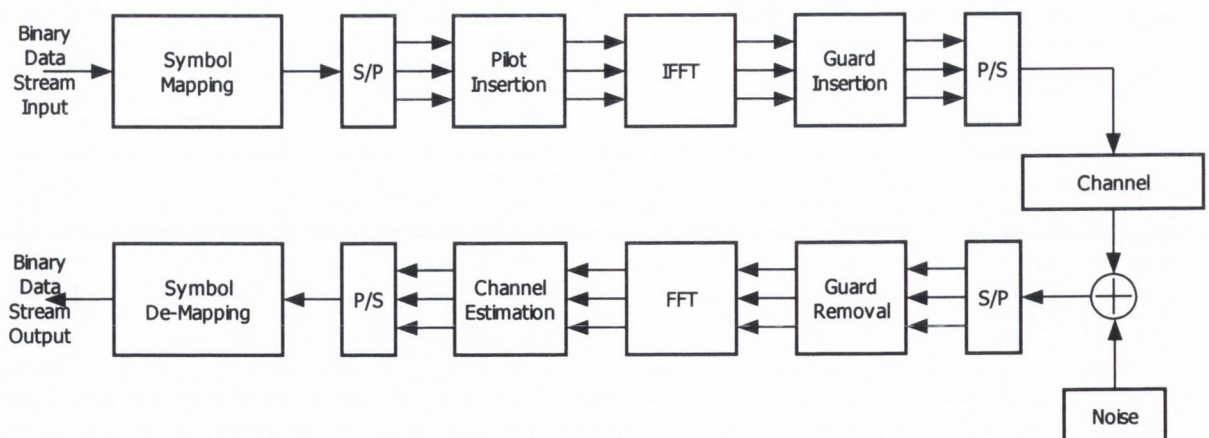


Figure 2.1: Block diagram of a typical OFDM transceiver.

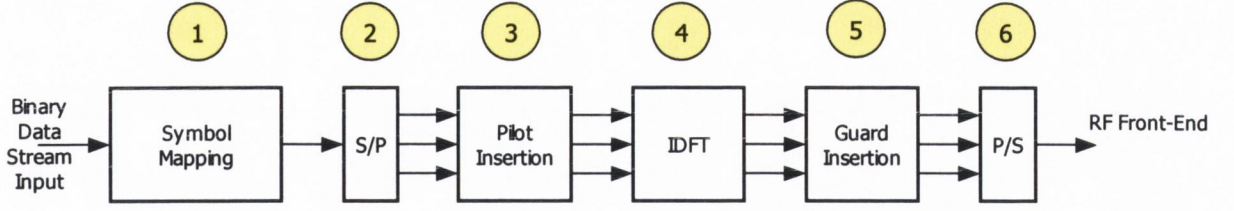


Figure 2.2: Block diagram of the OFDM generation process with the core processing steps labelled 1 to 6.

In Figure 2.1, the OFDM transmitter has six stages labelled 1 to 6. In order to preserve the clarity of the description of the core elements of an OFDM transceiver system, the interpolation, decimation, and associated stages of filtering required for converting the sampling rate of the signal sequences to the sampling rate of the analogue to digital/ digital to analogue converters (ADC/DAC) will not be included for the moment.

2.4.1 Stage 1 – Symbol Mapping

Stage 1 in the OFDM transceiver is the symbol mapping stage. The purpose of this stage is to map an incoming binary stream to a stream of modulated signals, or data symbols.

The term ‘data symbol’ is used extensively in this thesis therefore it is important to clarify its meaning before progressing with the description of the OFDM transceiver. A data symbol is a complex-valued number that represents a point on a constellation diagram. A constellation diagram is a two-dimensional diagram that represents the amplitude and phase of a signal point and shows how an information binary value may be mapped to a modulated signal. The x-axis denotes the real part of a complex-valued signal increasing from 0 at the origin to ∞ in the positive x-direction and decreasing from 0 at the origin to $-\infty$ in the negative x-direction. The y-axis corresponds to the imaginary part of the signal value and also passes through the origin, increasing from $0j$ to ∞j in the positive y-direction and from $0j$ to $-\infty j$ in the negative y-direction. The maximum absolute values of the signal points depend on the modulation scheme. For a simple scheme such as Binary Phase Shift Keying (BPSK), as shown in Figure 2.3, the maximum absolute value of a BPSK signal is typically chosen to be unity. A data symbol is the amplitude and phase value of a signal point on a constellation diagram. The constellation diagram has 2^N signal points depending on the modulation scheme being used, where N is the number of bits represented by each signal point.

Stage 1 therefore converts a sequence of binary values into complex-valued signals, where the original binary value is represented by the amplitude and phase of the signal point. In other words, Stage 1 performs the actual sub-carrier modulation. The number of binary values mapped to a data symbol depends on the modulation scheme in use. There are many different modulation schemes that may be employed in an OFDM transceiver. Binary Phase Shift Keying (BPSK) maps one bit to one symbol. Quadrature (or Quadriphase as it also known as) Phase Shift Keying (QPSK) involves the mapping of two bits to each symbol while 16-Quadrature Amplitude Modulation (QAM) maps four bits to each symbol. The signal point therefore may be one of the 2^N possible constellation points. An example of a typical constellation diagram that represents BPSK modulation is shown in Figure 2.3, and an example of a typical constellation diagram for QPSK is shown in Figure 2.4 and a typical 16-QAM constellation is shown in Figure 2.5. The signal points in red are all the possible signal point states for these examples. For the remainder of this chapter, when presenting examples, BPSK and QPSK modulation schemes are used. However, a very wide range of other suitable modulation schemes exist.

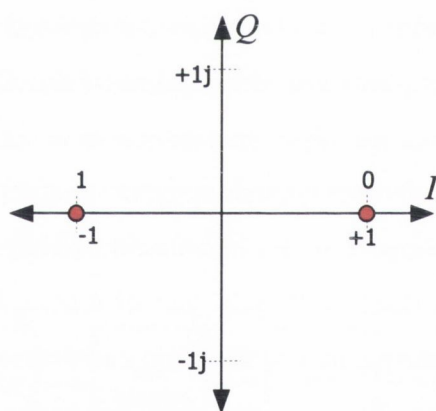


Figure 2.3: Example of a BPSK constellation diagram.

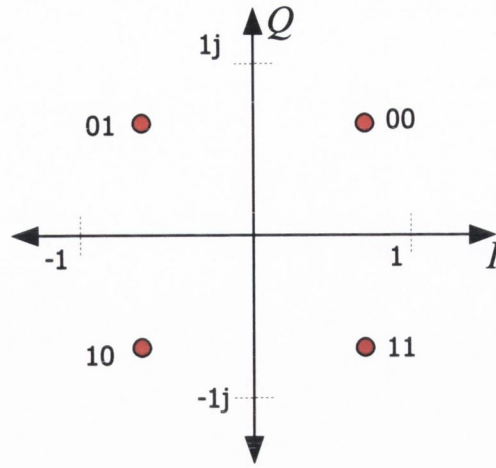


Figure 2.4: Example of a QPSK constellation diagram.

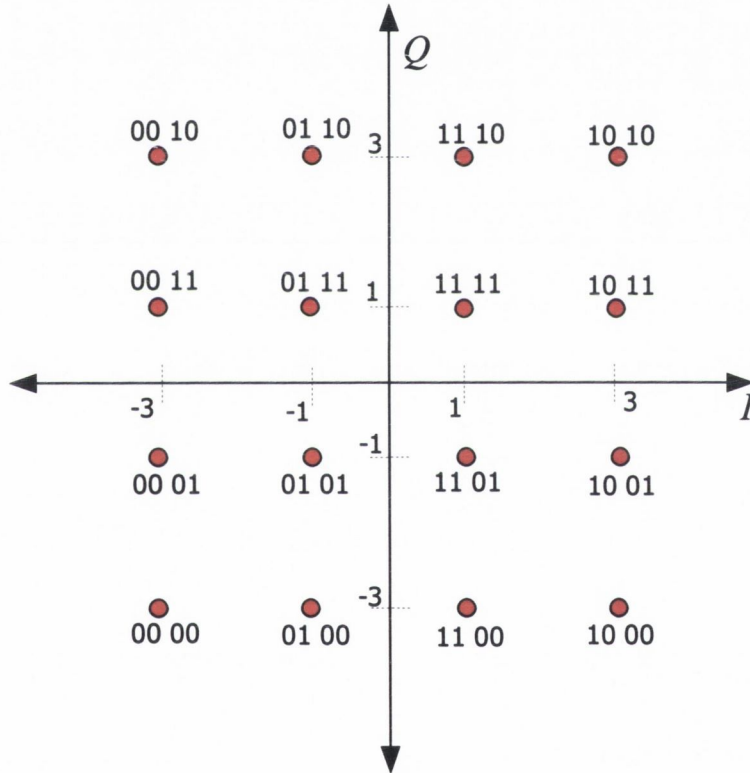


Figure 2.5: Example of a 16-QAM constellation diagram.

Returning to the OFDM transceiver the basis functions used in OFDM are sinusoids and are of the form:

$$\phi_n(t) = A(t) \exp(j2\pi f_n t) \quad [2.1]$$

where f_n is the frequency of the n -th sub-carrier, and $A(t)$ is the amplitude of the signal. The basis function described in Eq. [2.1] may be expressed in terms of amplitude and phase terms:

$$\phi_n(t) = A(t)\cos(2\pi f_n t) + A(t)j\sin(2\pi f_n t) = I(t) + jQ(t) \quad [2.2]$$

where $I(t)$ is known as the In-Phase or I component and $Q(t)$ is the Quadrature or Q component. Each sub-carrier may therefore carry data by representing the data bit in terms of a phase and/or amplitude variation. It is the resulting data symbol from the symbol mapping process that gives the value of the phase and/or amplitude of the basis functions.

As stated already a BPSK modulation represents one binary value at a time. It therefore has two possible phase values, $[0, \pi]$ radians, which are expressed in terms of In-Phase (I) and Quadrature (Q). A binary one may be chosen to correspond with $0, (2\pi)$ radians or the complex-valued signal point (symbol), $1 + j0$, and a binary zero is therefore represented as π radians or the complex-valued constellation point (symbol), $-1 + j0$, as shown in Figure 2.3. Likewise QPSK enables the transmission of two binary values, known as a dibit, at any one time. The set of possible dibits is $[00,01,10,11]$, which may be expressed as phasors described by the expression:

$$\phi(k) = A(k)\exp\left(\frac{2\pi k}{N}\right) \quad 0 \leq k < N \quad [2.3]$$

where $A(k)$ is the amplitude associated with the k -th symbol and N is the number of bits that may be transmitted (in this example, $N = 4$).

For the remainder of this thesis, the array of complex-valued data symbols will be expressed as :

$$c(k) = A(k)\exp\left(\frac{2\pi k}{N}\right) = a(k) + jb(k) \quad 0 \leq k < N \quad [2.4]$$

where N is the number of binary values represented as one data symbol, $c(k)$ is the k^{th} complex-valued symbol (or the data symbol associated with the k^{th} sub-carrier), $a(k)$ and $b(k)$ are the k^{th} In-Phase (I) and Quadrature (Q) values, respectively.

To summarise Stage 1; the binary stream is grouped according to the relevant number of bits to be mapped. The number of bits in the group depends on the modulation scheme chosen. Each group of bits (or bit in the case of BPSK) is mapped to a data symbol, $c(k)$. Figure 2.6 depicts this process for the BPSK case. In this example, a byte (eight binary values) from a sequence of binary data is converted to eight complex-valued signal points using BPSK as the symbol mapping scheme.

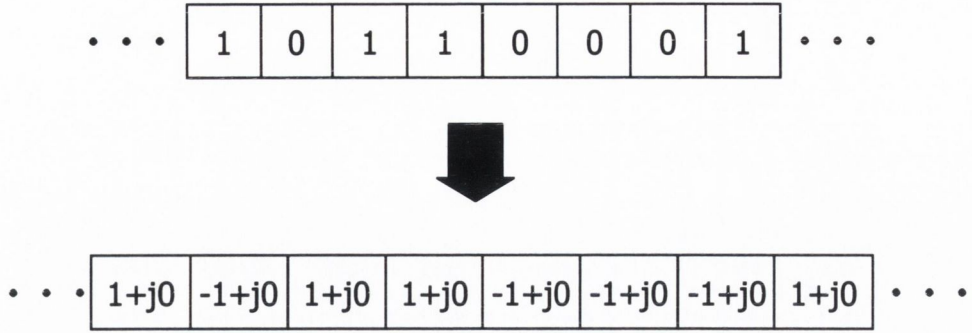


Figure 2.6: Mapping a sequence of binary values to complex-valued signal points using BPSK.

2.4.2 Stage 2 – Serial to Parallel Conversion

Once the sequence of binary values has been expressed in complex-valued (Cartesian coordinates) form, the next step is to prepare the data signal sequence for conversion to an OFDM waveform by converting the serial data symbol sequence into a number of shorter parallel sequences. Stage 2 rearranges the sequence of data symbols into a number of smaller sub-sets of data symbols. The size of the each sub-set (i.e., the number of data symbols that may be contained in each sub-set) is determined by the number of sub-carriers that will be used to finally transmit the data.

The term ‘sub-carrier’ is used throughout this thesis and refers to a single carrier frequency which is used to transmit information. OFDM uses several sub-carriers, which are multiplexed and may be used to transmit a large amount of information at any one time. OFDM uses sub-carriers which are arranged so that they do not interfere with one another. Each sub-carrier is referenced by a sub-carrier index, which is the FFT bin number used for the OFDM creation process. This topic is dealt with in more depth later in the chapter.

As one complex-valued signal point is transmitted for each sub-carrier used, then if N sub-carriers are used for transmission, each sub-set of signals contains N complex-valued signal points (or three symbols in this example). Continuing on from the example shown in Figure 2.6, suppose it is required to transmit eight complex-valued signal points using four sub-carriers, this means that each sub-carrier will transmit two complex-valued signal points (or two data symbols) as shown in Figure 2.7.

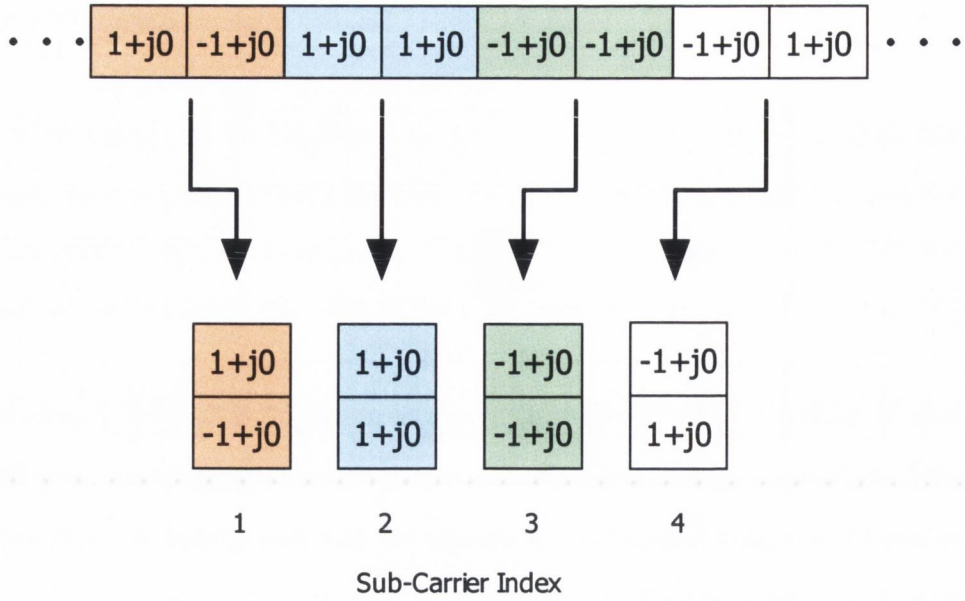


Figure 2.7: A sample of eight symbols from a symbol sequence assigned to four sub-carriers.

As can be seen in Figure 2.7, the set of 8 signal points must be converted to shorter sub-sets of signal values, each sub-set in this example containing two complex-valued signal points. The allocation of symbols to sub-carrier may be at the discretion of the transceiver designer, and in the example shown in Figure 2.8, the first two symbols that encounter the serial to parallel conversion stage, (Stage 2), are assigned to sub-carrier index 4 and the following two symbols are assigned to sub-carrier index 3. This process continues until sub-carrier 1 has been assigned two symbols. At this stage, the next eight symbols to arrive undergo the same sub-carrier allocation process beginning with sub-carrier index 1. This process repeats until all of the symbols required to be transmitted are assigned to sub-carriers.

Two OFDM symbols will be required to transmit the byte as originally shown in Figure 2.6 and again in Figure 2.7. The term OFDM Symbol is used very often in this thesis and worth defining carefully.

The OFDM Symbol is the time-domain waveform representation of the total number of multiplexed sub-carriers. The total number of sub-carriers depends on the FFT size used. An OFDM symbol may comprise information-carrying sub-carriers, pilot sub-carriers and even null carriers (sub-carriers with zero value). Each of the information-bearing sub-carriers in one OFDM symbol contains one complex-valued data symbol. Other available sub-carriers may be used for allowing the receiver to estimate the effects of the

communications channel, or may contain no information (i.e., the data symbol associated with that sub-carrier is zero). These are discussed in later sections.

The OFDM symbols are illustrated conceptually in Figure 2.8. OFDM symbols are separated from one another by either a null signal or non-information bearing signals for a short time interval called a guard period (or interval). A guard period and is used to allow the effects of the previously transmitted OFDM symbol to be sufficiently attenuated so that it does not interfere with the next OFDM symbol that is transmitted following this.

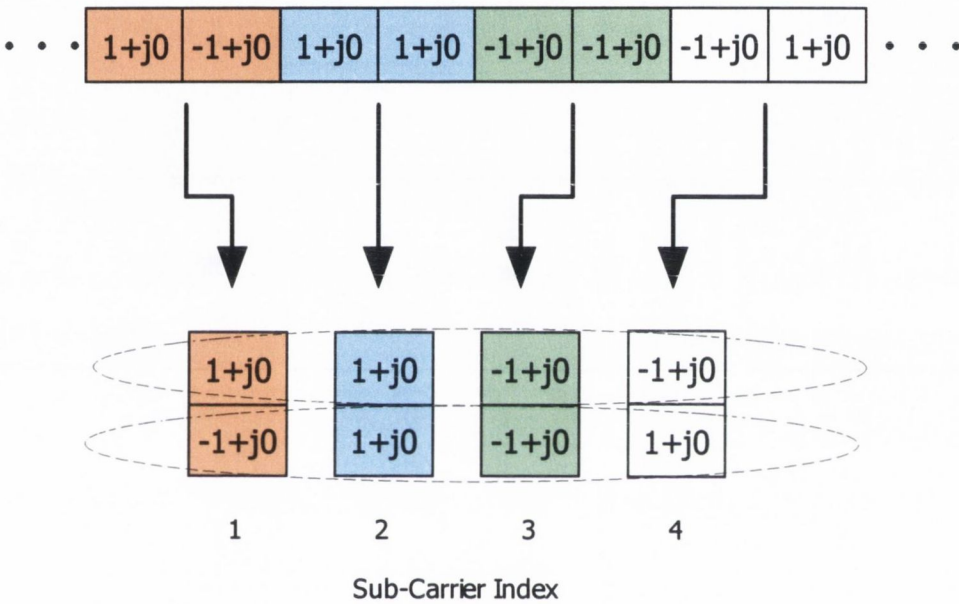


Figure 2.8: OFDM Symbol Generation.

2.4.2.1 Symbol Interleaving

Symbol interleaving is also an inherent function of Stage 2. Symbol interleaving is a process of re-ordering symbols in such a way that the end result is a non-sequential new arrangement of the original symbols. The symbol values are not modified but the order in which they occur in time (and frequency) is changed. Symbol interleaving has the attractive property of enabling the effects of a local deep fading null to be averaged over the entire set of transmitted OFDM symbols thus improving the robustness of the transmitted signal.

The manner in which the symbols are assigned to the sub-carriers in Stage 2 of the OFDM transmitter process illustrated in Figure 2.2 means that the symbols are interleaved in time

and frequency. The symbols in Figure 2.7 are interleaved in frequency as every two-symbol group in the eight-symbol example is assigned to a different sub-carrier index. The symbols are interleaved in time also as the first symbol in the two-symbol grouping in the example shown in Figure 2.8 is assigned to each sub-carrier first. This means that the resulting sequence of four-symbols, each symbol assigned to a different sub-carrier, will eventually form the first OFDM symbol. This OFDM symbol will be transmitted before the second sequence of four symbols meaning that the time-varying effects of the communications channel on the sequence of transmitted OFDM symbols may have changed and therefore may not affect the original sequence of symbols in the exact same way.

The main advantage of this interleaving process is that it increases the resilience of the transmission method to noisy and/or fading channel environments. If for example (using Figure 2.8 as the example case), the waveforms are transmitted through a communications channel subject to fading, the first OFDM symbol that appears on the channel is attenuated sufficiently enough to result in the unrecoverable loss of this OFDM symbol. This means that only every second consecutive symbol in the original sequence of symbols prior to the interleaving process is lost instead of four consecutive symbol losses (or errors). The correction of four consecutive errors requires a longer and more robust error correction scheme therefore symbol interleaving increases the resilience of OFDM to the effects of channel disturbances on the transmitted waveforms.

2.4.3 Stage 3 – Pilot Carrier Insertion

The example shown in Figure 2.7, describes the sub-carrier allocation scheme using only four sub-carriers. In reality, many more sub-carriers may be used to increase the number of symbols that may be transmitted at any one time. Certain sub-carriers may be used, not as information-carrying carriers but as reserved sub-carriers that aid the receiver to equalise the effects of the channel on the received signal. These reserved sub-carriers are called pilot carriers and the normal information-carrying sub-carriers may be either interspersed with these pilot carriers or the pilot carriers may comprise an entire OFDM symbol.

Pilot carriers are used in OFDM transceiver systems to enable the receiver to correct any frequency and timing errors that may cause misalignment between the transmitter and

receiver. These special carriers may also be used to allow the receiver to estimate where the sequence of OFDM symbols begins in an intercepted received signal sequence. Predetermined symbols are transmitted using these pilot sub-carriers and then the received signal values are compared with the known transmitted symbols. A pilot symbol is a complex-valued number representing a point on a constellation diagram. A pilot symbol is usually chosen from one of the signal points furthest away from the origin of the constellation diagram in order to ensure that it is a high-power pilot symbol. Pilot symbols with a high power may be chosen from the constellation diagram of a 16-QAM, 64-QAM, or even 256-QAM. Using a pilot symbol with a high power both differentiates it from a normal information-bearing OFDM symbol as well as increasing the chances of the receiver actually receiving it.

Figure 2.9 takes the original sub-carrier allocations used in Figure 2.8, spaces them apart by one sub-carrier, and then one pilot symbol is inserted into the sub-carrier index between each original information-carrying sub-carrier index. This is one method of pilot-carrier insertion and this is used to illustrate the concept described in this section. The number of required sub-carriers has increased from four to seven, although only four sub-carriers are carrying the information from the selected byte in the sequence as shown in Figure 2.6. The data symbols are separated in order to clarify the order in which the completed OFDM symbols will be transmitted. Both of the rows of sub-carrier values will be converted to OFDM symbols in parallel. However, the bottom row of data symbols and pilots denoted OFDM Symbol 1 in Figure 2.9 will be the first OFDM symbol to be transmitted when the waveforms are converted to a serial sequence of OFDM symbols.

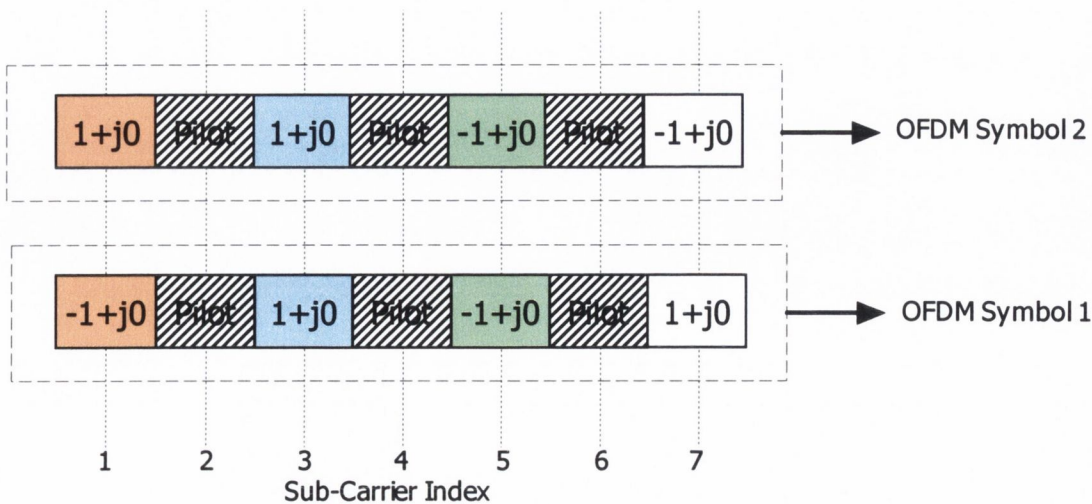


Figure 2.9: The information-bearing sub-carriers are interspersed with pilot sub-carriers and prepared for the IFFT stage where the OFDM symbols are created.

2.4.4 Stage 4 – Inverse Fast Fourier Transform (IFFT)

Stage 4 is used to create OFDM symbols from the frequency domain waveforms by calculating the inverse Discrete Fourier Transform (DFT) of the input OFDM waveforms. The Fast Fourier Transform (FFT) algorithm developed by Cooley and Tukey [Cooley1965] [Cooley1967] reduces the number of computations required to perform a DFT. Therefore the FFT and Inverse FFT (IFFT) are chosen for the OFDM transceiver system presented in this thesis. OFDM symbols, which are time-domain representations of the multiplexed OFDM sub-carriers are obtained using the IFFT. This process is described in this section.

A single-carrier transceiver uses one RF oscillator, which is a sinusoidal wave oscillating at the frequency of interest. Multi-carrier modulation requires two or more oscillators which are combined to form the transmission system. The IEEE 802.11a example referred to in Stage 3 would require fifty-two unique oscillators, all overlapping each other but not interfering with any of the other oscillator centre frequencies. Multiple closely-spaced RF oscillators may result in RF-mixing due to the possibility of RF-leakage from adjacent oscillators in addition to a more complex and larger transceiver system than a digital equivalent. Small variations in the RF oscillator frequencies may also result in loss of orthogonality between the carrier frequencies.

To overcome the need for a large number of unique RF oscillators, Stage 4 uses the Fast Fourier Transform (FFT) algorithm to generate the combined set of sub-carriers, which are multiplexed whilst maintaining orthogonality between the sub-carriers, and eliminating the added complexity and physical space requirements for one analogue oscillator per sub-carrier. The IFFT is a graceful and efficient method of producing an OFDM symbol, which is real-valued and comprises the multiplexed sub-carriers, which are converted to a time-domain waveform. Before going in to the details of the IFFT it is important to look at the idea of orthogonality in more detail.

2.4.4.1 Orthogonality

Orthogonal signals are carrier frequencies that do not interfere with each other. The spectrum of each sub-carrier may overlap all the others but the centre frequency of each

sub-carrier occurs at a null in the spectrum of all the other sub-carriers. In OFDM, the carriers may be spaced as close as theoretically possible to the rest of the set of carriers thus resulting in very high spectral efficiency. These orthogonal frequencies may be transmitted across a single communications channel, and at the receiver, they may be demodulated as independent signals without suffering from signal degradation resulting from interference from adjacent carriers if signal orthogonality is maintained.

Temporal orthogonality is an inherent feature in schemes such as Time Division Multiple Access (TDMA) as only one source out of a possible set of multiple sources is transmitted per timeslot. Non-OFDM transmissions maintain orthogonality by ensuring that the separate sources are spaced far enough apart in the frequency domain to ensure that no interference occurs. OFDM however, maintains orthogonality even though the carriers are as densely packed as possible by ensuring that all the sub-carrier baseband frequencies are integer multiples of the reciprocal of the symbol period. The symbol period is dependent on the sampling (chip) rate of the transmitter. This means that all the sub-carriers have an integer number of cycles per symbol. In other words, in a particular time interval T , each sub-carrier frequency is an integral multiple of a base frequency, and the number of cycles between two adjacent sub-carriers differs exactly by one. For example, the IEEE802.11a standard uses a 20 MHz sampling rate and the symbol period is $3.2\mu S$ [IEEE802.11a].

Two signals are orthogonal if the dot product of the two is zero. This means that if two signals are multiplied and summed over an interval a to b as described by Eq. 2.5, then for an orthogonal set of signals, the result is zero. Suppose we have a set of signals Ω , where Ω_p is the p -th element in the set.

Orthogonality exists if:

$$\int_a^b \Omega_p(t) \Omega_q^*(t) dt = \begin{cases} K & \text{for } p = q \\ 0 & \text{for } p \neq q \end{cases} \quad [2.5]$$

where $[a, b]$ is one symbol period and $*$ denotes the complex conjugate of the signal.

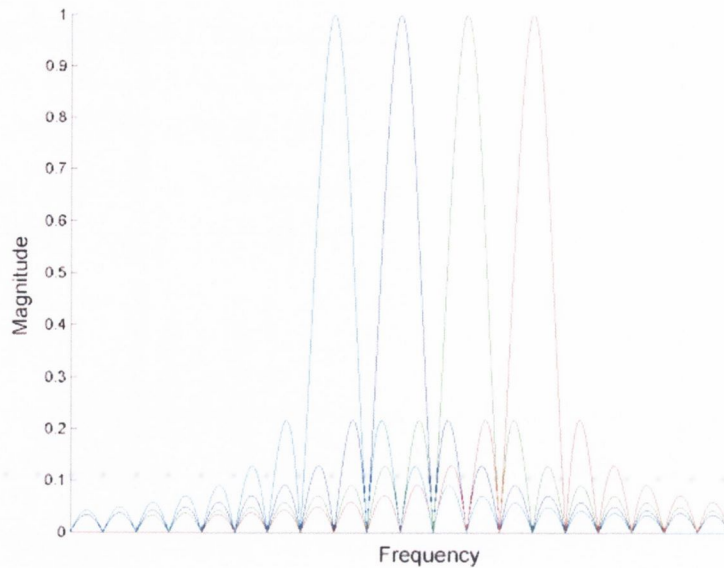


Figure 2.10: Power spectrum of four-carrier OFDM spaced two carriers apart.

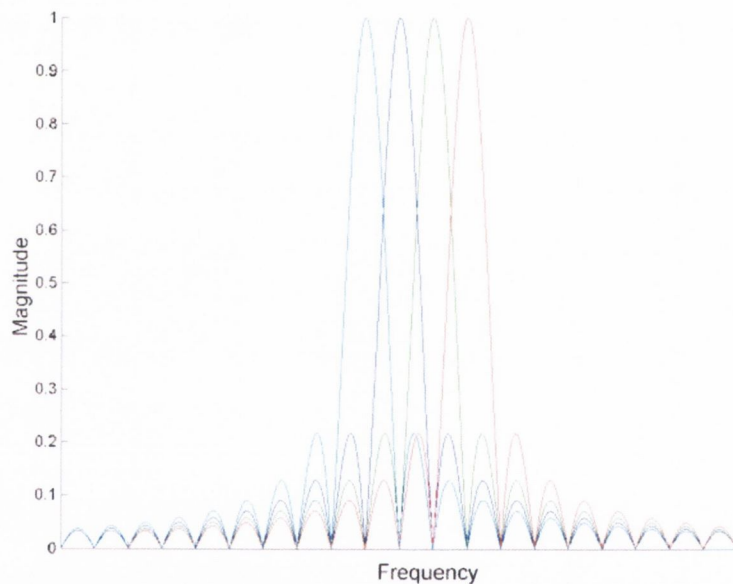


Figure 2.11: Power spectrum of four-carrier OFDM spaced one carrier apart.

Figure 2.10 shows the power spectrum of four sub-carriers spaced apart by two sub-carriers. This spacing is chosen as it is easier to see that the centre frequency of each of the sub-carriers occurs at a null in the spectrum of the three other carriers. Carrier spacing is chosen by simply choosing which FFT bin number to insert the complex-valued signal point. For example (using the example illustrated by Figure 2.9), in order to space the sub-carriers apart by one carrier, the complex-valued signal points may be placed in FFT bins one, three, five, and seven. Figure 2.11 is the power spectrum of four carriers with a spacing of one carrier. These carriers are packed as close as possible together, whilst still maintaining signal orthogonality. This shows that OFDM is a highly spectral-efficient system and hence is often regarded as the optimal version of multi-carrier transmission

schemes. Although only four carriers are illustrated, the number of carriers in a real system is generally much higher and typically spaced one sub-carrier apart. For example the total number of sub-carriers in 802.11a wireless LAN is 52 while for Digital Video Broadcast-Terrestrial (DVB-T), up to 6817 sub-carriers can be employed.

Multiple coherent modulators can be implemented simply and efficiently using the IFFT of the parallel data stream. By exploiting the properties of the FFT, sub-carrier frequencies are chosen so as there are an integral number of cycles in a given symbol period resulting in signal orthogonality. The amplitude and phase of the sinusoid, which are used to represent symbols, do not affect the sub-carrier orthogonality.

Each sub-carrier has a $\sin(x)/x$ spectrum because the transmitted pulse is effectively a rectangular-windowed sinusoid. Provided that the sinusoid has the correct sub-carrier frequency and a very small bandwidth, each carrier will have a spectral null at the centre frequency of all the other carriers resulting in zero Inter-Carrier Interference (ICI). Relative to the symbol rate, the OFDM carriers occupy a significant amount of spectrum. This characteristic is not problematic given the carriers overlap each other and it is this carrier overlapping combined with the overall data rate that results in this scheme's high spectral efficiency. OFDM is highly reliant on maintaining time and frequency synchronisation in order to prevent ICI and allow correct demodulation of each sub-carrier. Each sub-carrier occupies a very small segment of the total bandwidth used for transmission therefore the frequency-selective properties of the channel may be considered to be uniform across the very small sub-carrier bandwidth. A single-carrier transmitter may use a significantly larger bandwidth necessitating a more complex equalisation stage as the channel effects may be frequency-selective. Cimini [Cimini1985] showed that equalisation of received OFDM signals is therefore much simpler than for single channel transceiver systems

The length of an OFDM signal is dependent on the size of the FFT used. A FFT of length N_{FFT} , may use N_{SC} sub-carriers, where

$$N_{SC} = N_{FFT} / 2 \quad [2.6]$$

This represents the frequency domain from 0 to $F_S / 2$, where F_S denotes the sampling frequency.

The bandwidth of the each sub-carrier is $1/N_{FFT}T$, where the symbol period is denoted by T and $T = 1/F_s$.

The OFDM waveform in the frequency domain (before conversion to a time-domain OFDM symbol) may be created digitally by representing the Real and Imaginary sub-carriers, of total length, N_{FFT} as an array of N_{FFT} complex-valued numbers. The N_{SC} Real sub-carriers, $c(0) \cdots c(N_{SC} - 1)$, may then be inserted into the array indices extending from array index $(N_{FFT} / 2) - 1$, representing zero Hz (DC), to array index $N_{FFT} - 1$. The N_{SC} Imaginary sub-carriers, which are the complex-conjugate of the Real sub-carriers may then be inserted array indices 0 to $N_{SC} - 1$.

Progressing from the general form of the OFDM generation process expressed in Eq. [2.3], the real-valued baseband OFDM signal may be generated by performing an Inverse Fast Fourier Transform (IFFT) on the completed array of N_{FFT} complex-valued data/pilot symbols denoted by $X(k)$, where k denotes the k^{th} element of the array, as follows:

$$s(n) = \frac{1}{\sqrt{N_{FFT}}} \sum_{k=0}^{N_{FFT}-1} X(k) \exp\left(\frac{j2\pi nk}{N_{FFT}}\right), \quad 0 \leq n \leq N_{FFT} - 1 \quad [2.7]$$

where the scaling factor $\frac{1}{\sqrt{N_{FFT}}}$ is used to preserve Parseval's theorem.

2.4.4.2 Practical implementation

In order to correctly create an OFDM symbol, the OFDM signal is created in the frequency domain first. As this is implemented digitally, the sub-carriers are 'inserted' into the FFT bin indices corresponding to the desired sub-carrier frequency. Sub-carrier insertion is the process of assigning a complex-valued data symbol to a particular FFT bin number. Each bin in the FFT array corresponds to a unique carrier frequency, and for maximally packed sub-carriers, the sub-carrier are spaced apart by one carrier frequency. In other words, there are no empty FFT bins between sub-carriers. Assuming a perfect communications channel and a perfect receiver, then the originally transmitted data symbol may be extracted at the receiver by accessing the same FFT bin number used to transmit the data symbol.

The frequency spectrum consists of real valued components and imaginary frequency components extending from $-\infty$ to $+\infty$ Hz. In this practical case, the number of frequencies is limited by the FFT size. In order to correctly create the time-domain waveforms, the conjugate of the real sub-carrier values (the imaginary frequency components) must be inserted into the FFT array also. This is shown in Figure 2.12, where four sub-carriers are being used for data transmission, referenced by sub-carrier indices 1, 3, 5, and 7. Pilot symbols are inserted in the sub-carrier indices 2, 4, and 6, which are shaded in the figure to highlight the difference between them and the sub-carriers containing data symbols. In this example, a twenty-two bin FFT is utilised meaning that a maximum of eleven sub-carriers may be used for data transmission.

The example shown in Figure 2.12 uses more carriers than is necessary in order to improve the clarity of the description. An increased number of sub-carriers results in a longer time-domain signal sequence however. In this thesis, the minimum size of the FFT is chosen to be $2N_{SC}$, where N_{SC} is the number of sub-carriers chosen by the designer. This size accounts for the conjugate sub-carriers (imaginary frequencies) that must also be present in order to perform the IFFT. An extra sub-carrier is also used to represent zero frequency as shown in Figure 2.12. Usually the FFT size is an integral multiple of two (or modulo 2) in order to reduce the complexity and FFT processing time.

The complex conjugates of these sub-carriers must therefore be inserted in the FFT array indices 15, 17, 19, and 21 to create a mirror-image of the real sub-carriers. Referring to Fig. 2.9, the data symbols which are denoted by the colours orange, blue, green and white correspond to the sub-carriers shown in Figure 2.8. The vertical dashed line in Figure 2.12 represents zero Hz, or DC, with the imaginary frequencies increasing towards the left and increasing real frequencies towards the right.

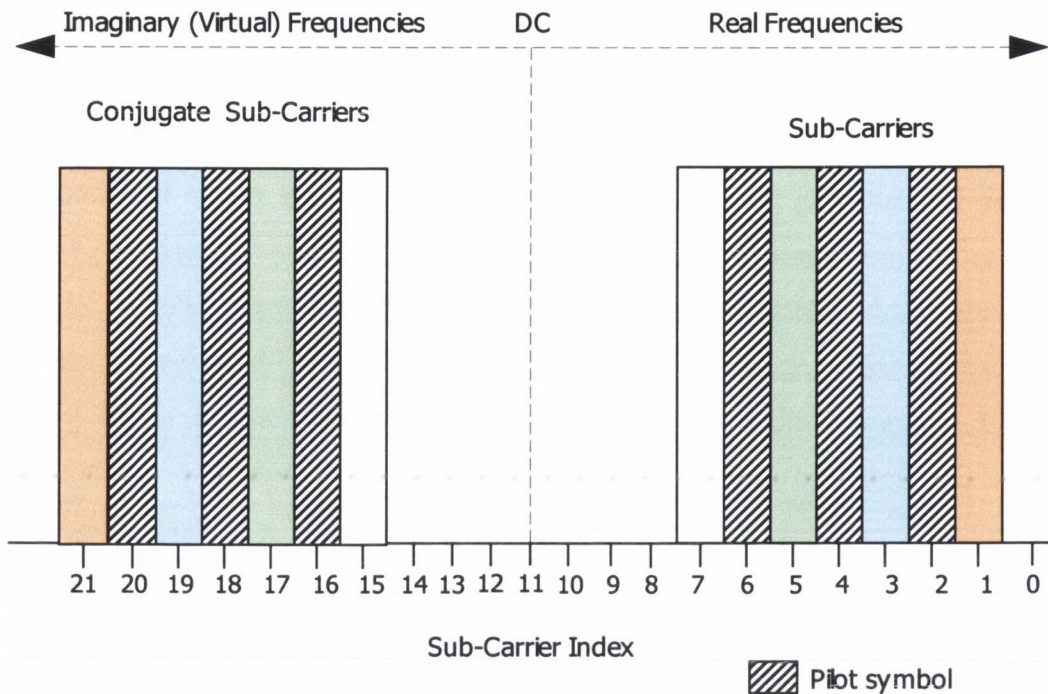


Figure 2.12: Practical representation of the real and imaginary sub-carrier array using a 16-bin FFT prior to the IFFT processing stage.

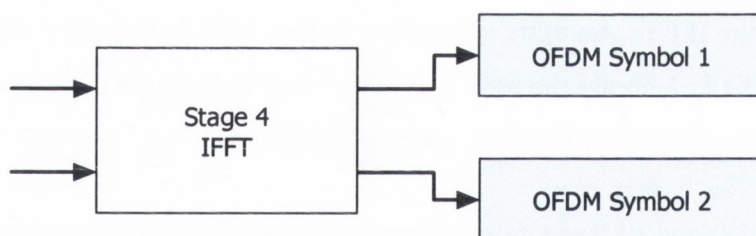


Figure 2.13: Two OFDM symbols created in parallel.

The IFFT stage accepts a complex-valued signal sequence and the resultant carrier frequency is determined by the sampling rate of the transceiver and the IFFT bin number that the data symbol is present on. The end-product of Stage 4 is an array of OFDM symbols, which are real-valued time-domain waveforms. These are in parallel and therefore they must be converted to a serial sequence of OFDM symbols before they can be transmitted.

2.4.5 Stage 5 –Guard Period Insertion

Referring to Stage 5 in Figure 2.2, the main objective of the guard interval (or period) insertion stage is to help maintain sub-carrier orthogonality by reducing possible inter-symbol interference (ISI) effects. ISI is interference to the current OFDM symbol present on the communications channel due to one or more delayed replicas of previously

transmitted OFDM symbols that have not been fully attenuated by the time the next OFDM symbol is transmitted. A guard period is appended to the OFDM symbol to ensure that the effects of the previously transmitted symbol have time to dissipate completely before the next symbol is transmitted. As the transmitted signal passes through a multi-path fading channel, several reflected versions of the original transmission may at the receiver at slightly different instances depending on the length of the propagation path of each version (and thus resulting in a longer propagation time) resulting in a spread of delayed and possible reduced amplitude versions of the single transmitted signal resulting in ISI and possible loss of orthogonality.

Frequency and timing offsets manifest themselves as differences in phase and amplitude between the received and known transmitted data symbols allowing the receiver to estimate and correct the offsets in order to lock on to the correct frequency. In addition, the transmitter may transmit pilot carriers preceding the main body of the OFDM waveform which contains information from the source(s). The OFDM receiver can then begin extracting the data contained in the received OFDM signal when this signal (a series of known pilot carriers) is received. Instead of, or in conjunction with comparing a known data sequence with a received sequence, pilot carriers may also be chosen to have much greater power than the normal data sub-carriers. These are known as *boosted* pilot carriers and are created by choosing signal points from a higher-order signal constellation than the data sub-carrier modulation scheme such as 16-QAM (as shown in Figure 2.5), 64-QAM or even 256-QAM. These high power pilot carriers are simply points on a constellation diagram that are furthest from the origin. The receiver may be designed to begin its data extraction process when the received power exceeds a threshold limit indicating that the boosted pilots may have been transmitted. The use of pilot carriers adds an extra overhead to the transceiver as they do not carry data from the source and reduce the amount of spectrum available for valid data transmission if dedicated pilot sub-carriers are used. An example of this is the IEEE 802.11a standard, which uses OFDM. Four dedicated pilot sub-carriers are used out of a total of fifty-two sub-carriers; this results in forty-eight remaining sub-carriers for data transmission. A more detailed description of the IEEE 802.11a standard will be given in Chapter 3.

An OFDM symbol has a useful period of duration of N_{FFT} samples and this OFDM symbol may be preceded by the guard interval with a duration of N_{GI} samples, which is longer than the expected Channel Impulse Response (CIR) in order to minimise the

possibility of ISI. The total length of the OFDM symbol including the guard-interval is therefore $N_{FFT} + N_{GI}$ samples.

Figure 2.14 shows the two OFDM symbols obtained from Stage 4, which have been converted to a serial sequence of OFDM symbols in order to describe the guard period insertion stage more clearly. The guard interval in this example is a simple null signal.

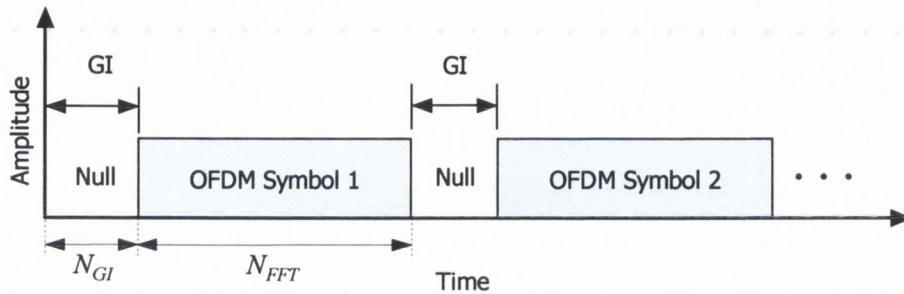


Figure 2.14: OFDM symbols in the time domain separated with a null symbol guard interval (GI).

The guard interval may be simply a zero-amplitude signal, or null signal, as shown in Figure 2.14, with duration greater than the CIR in order to allow the effects of the previously transmitted OFDM signal to dissipate to a negligible amount and therefore result in minimal interference between the previously transmitted symbol or number of OFDM symbols and the latest OFDM symbol.

2.4.5.1 Cyclic prefix

ISI and Inter-Carrier Interference (ICI) effects can be greatly reduced by pre-pending a cyclically extended part of the OFDM symbol to the actual symbol before transmission.

Instead of a null guard interval, the last N_{GI} samples of the OFDM symbol may be pre-pended to the start of this symbol resulting in a cyclic extension of the OFDM symbol as shown in Figure 2.5. This type of guard interval is known as a cyclic prefix and the OFDM symbol is therefore extended to an OFDM symbol of length $N_{FFT} + N_{GI}$ samples.

A cyclic prefix is an important feature of OFDM and is used to combat ISI and ICI effects introduced by the multi-path channel through which the signal is propagated. This results in a continuous, cyclically-extended signal at the seam of two serialised OFDM symbols,

as illustrated in Figure 2.16, thanks to the periodic nature of the FFT. The duration of the guard period, T_G , (corresponding to N_{GI} samples) should be longer than the worst-case delay spread of the multi-path channel in question. At the receiver, the demodulator will effectively ‘see’ an incoming OFDM symbol with an integral number of cycles within the FFT window, thanks to the cyclic prefix. Therefore, orthogonality will be preserved and ICI should not occur. A guard interval reduces data throughput as a percentage of the total received time-domain OFDM signal is not used for data transmission. The sampling instant, T_X , is chosen such that $\tau_{\max} < T_X < T_G$, where $\tau_{\max}$ is the maximum delay spread of the channel. Provided this condition is satisfied, there will be no ISI as the effects of the previously transmitted symbol will only be present on the channel for received signal samples within the range $[0, \tau_{\max}]$.

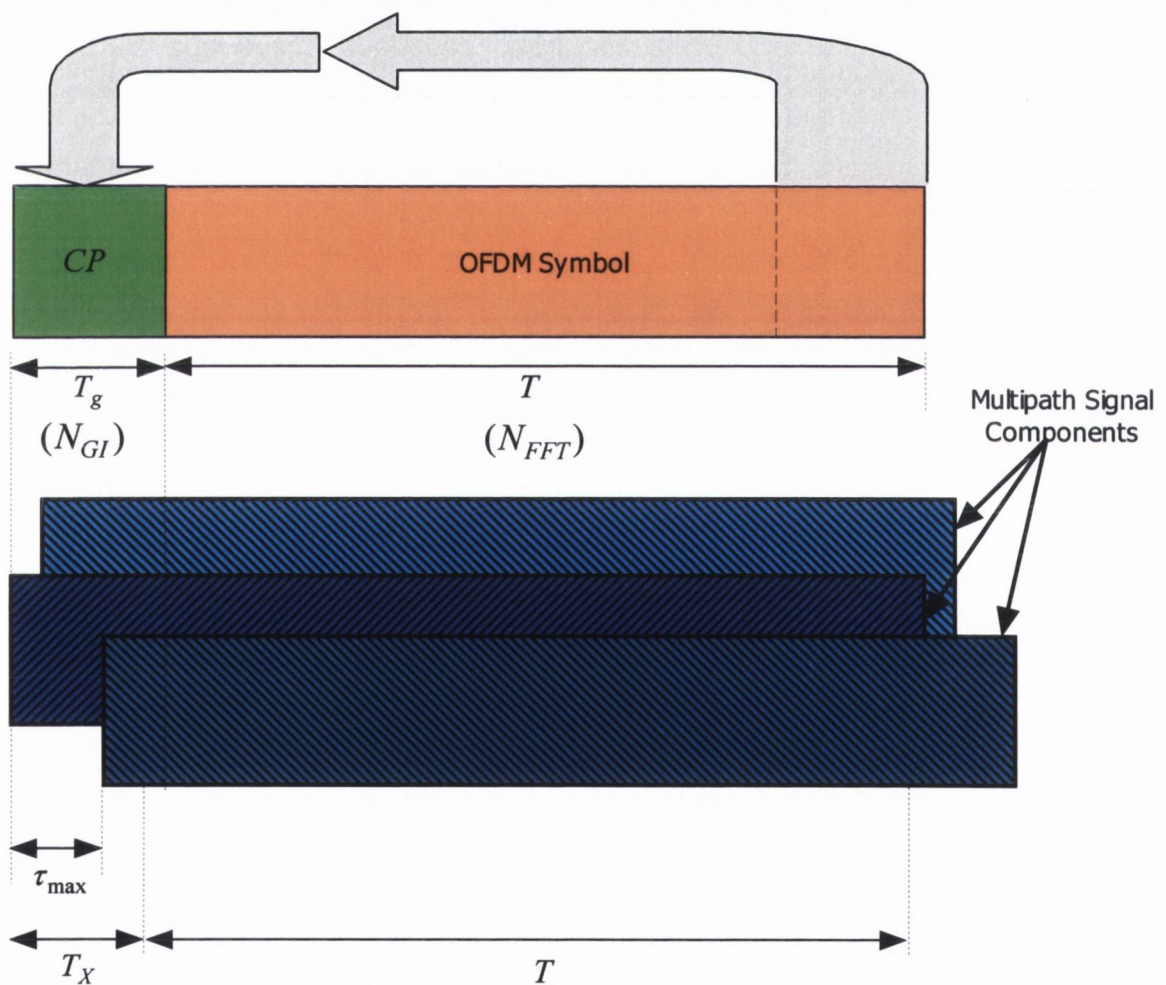


Figure 2.15: OFDM symbol pre-pended with a cyclic prefix and an illustration of how it helps to protect against ISI in a multi-path channel environment.

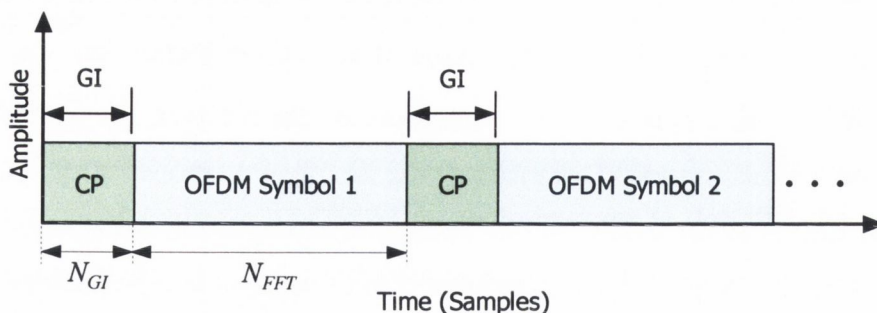


Figure 2.16: Time domain representation of a sequence of OFDM symbols with cyclic prefix (CP) guard intervals.

2.4.6 Stage 6 – Parallel to Serial Conversion

Stage 6 as shown in Figure 2.2, is where the parallel modulated OFDM symbols with pre-pended guard periods. The individual OFDM symbols, pre-pended with guard intervals are then converted to a serial sequence of OFDM symbols, which comprises the real-valued baseband OFDM waveform. The resulting OFDM signal structure is also illustrated in Figure 2.16 showing the sequence of OFDM symbols separated by guard intervals. This complete OFDM signal is then ready for up-conversion to the band of interest. As mentioned previously, OFDM modulates a sequence of binary data at a time and the total length, X_{len} , of the sequence of OFDM symbols after they have been converted from OFDM symbols of length, $N_{FFT} + N_{GI}$ samples, in parallel of to a serial sequence of waveforms is equal to:

$$X_{len} = M(N_{FFT} + N_{GI}) \quad [2.8]$$

where M is the number of OFDM symbols, N_{FFT} is the duration of the OFDM symbol before a guard interval is pre-pended, and the length of the guard interval is denoted by N_{GI} (i.e. the number of samples comprising the guard interval). This complete signal sequence, of length X_{len} , is referred to as an OFDM frame in this thesis.

2.4.6.1 OFDM Frames

An OFDM waveform may be generated on a frame by frame basis. An OFDM Frame is also an important concept and as such is defined here.

An OFDM Frame, (or simply, frame, in this context), is a sequence of OFDM symbols which are separated by a small time interval. An example of an OFDM frame is shown in Figure 2.17. This figure shows one OFDM frame and how OFDM symbols are ordered in relation to time and frequency. The guard interval between OFDM symbols is a cyclic prefix in this case. The size of the OFDM frame (i.e., the number of OFDM symbols that comprise the OFDM frame) is dependent on the amount of data that is required to be transmitted. This value is decided by the transceiver designer or by the radio user. Separating each OFDM frame is another time-interval called a frame guard. This frame guard is longer than the guard interval between the OFDM symbols and is used to enable the receiver to synchronise with the transmitter, which begins processing the received data upon detection of the start of the OFDM frame. This interval may also be used to enable the receiver to correct frequency offsets and initial channel equalisation.

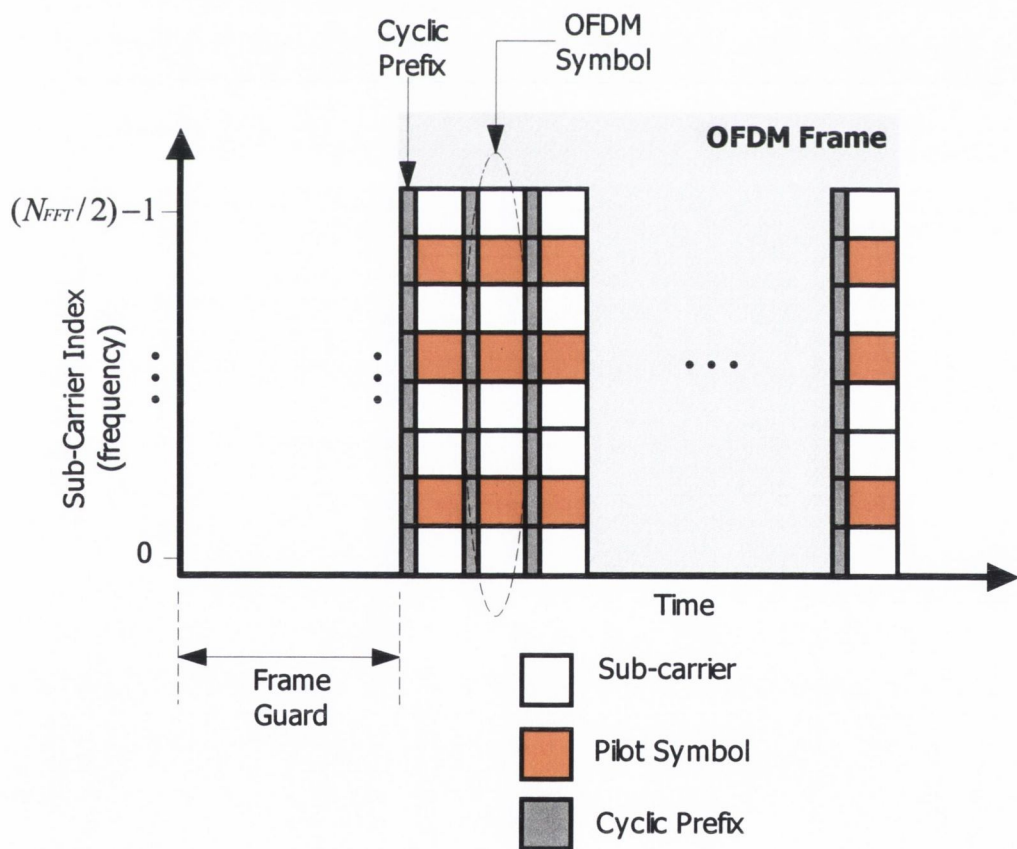


Figure 2.17: Time vs. Frequency view of an OFDM frame.

The term data payload is used in this thesis to denote the valid data portion of the OFDM frame. The valid data portion of a frame comprises the OFDM symbols contained in the frame that are carrying data and are not OFDM symbols used specifically for equalisation

and synchronisation. For example, the first two OFDM symbols in an OFDM frame may carry configuration or timing information whereas the rest of the OFDM symbols within the frame relate to the information from the source(s) such as sound, images or other data.

2.5 Up-Conversion

The final stage of the OFDM transmission process is to frequency-translate the baseband OFDM signal to the band of interest. Up-conversion is the process where a baseband signal is mixed with an RF signal oscillating at the desired pass-band centre frequency. In order to transmit this signal, this base-band signal must be mixed with the signal from an RF oscillator. For transmission in the Industrial, Scientific, and Medical (ISM) band, this RF signal may be generated using a 2.45 GHz reference RF oscillator. Usually, the frequency spectrum is windowed using a spectrum mask that effectively limits the spectral content of the signal in order to comply with either the radio designer's objectives or international standards. An example of this is the spectrum mask requirements for DAB. Signatories of the Terrestrial-DAB (T-DAB) spectrum-usage agreement at the 1995 European Conference of Postal and Telecommunications Administrations (CEPT) T-DAB planning meeting, agreed to abide by spectrum masks for terrestrial DAB out-of-band emissions. In this case, a suitable spectrum-mask was agreed upon and must be applied to the DAB signal before transmission [CEPT95]. The up-converted OFDM signal is then linearly amplified (or amplified in conjunction with the up-conversion process) and transmitted.

2.6 OFDM Demodulator

Neglecting channel effects for the moment and assuming perfect reception of the transmitted OFDM signal stream, the focus now turns to the main stages required for an OFDM receiver as shown in Figure 2.18. This particular OFDM receiver configuration as shown in this figure has been chosen as it best suits the design and objectives of this thesis. Extra necessary stages such as filtering, decimation, and frame/symbol synchronisation have been neglected for the moment. This has been done to expose the core elements of the receiver structure.

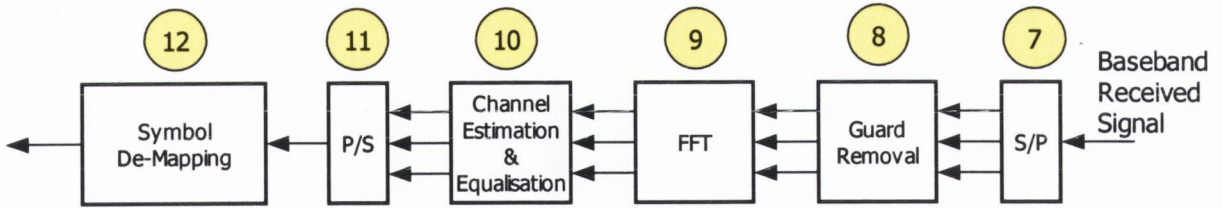


Figure 2.18: Basic block diagram of an OFDM receiver.

2.6.1 Stage 7 – Serial to Parallel Conversion

The received OFDM waveform is a continuous sequence of signal values. In order to reverse the process of transmission, this serial sequence of received signals must be partitioned into shorter OFDM waveform signals sequences in order to be demodulated. This is the reverse of the process carried out in Stage 6 in Figure 2.2.

Stage 7 is used to prepare the received OFDM signal for the receiver processes that will recover the data symbols from the sub-carrier indices and then produce the binary values associated with those data symbol values. It is assumed in this stage that the intercepted signal has been down-converted to base-band with a reference oscillator signal that has the exact same frequency and phase as the up-conversion oscillator. It is also assumed that the receiver is synchronised in time with the transmitter and has correctly estimated where the start of the OFDM frame occurs in the intercepted signal. The serial to parallel conversion stage shown as Stage 7 in Figure 2.18 converts a received OFDM frame, which is a real-valued signal sequence of length X_{len} as defined in Eq. [2.8], and translated in frequency to a baseband signal, into M parallel streams, where M is the number of OFDM symbols contained in the frame.

2.6.2 Stage 8 – Guard Period Removal

This stage simply removes the guard period which was originally pre-pended to each OFDM symbol in Stage 5 of the transmitter description. The M parallel real-valued baseband OFDM signals received still have either a null guard interval or cyclic prefix appended to the original OFDM symbol. Assuming perfect frame timing i.e. the receiver has correctly estimated where the start of the useful part of the OFDM symbol begins, the receiver may then eliminate the guard-period and proceed to the next stage of the demodulation process. The OFDM symbol extraction process simply neglects the first

N_{GI} number of samples in each parallel received OFDM signal where N_{GI} is the length of the guard interval defined in Eq. [2.8]. The function of this stage is illustrated in Figure 2.19.

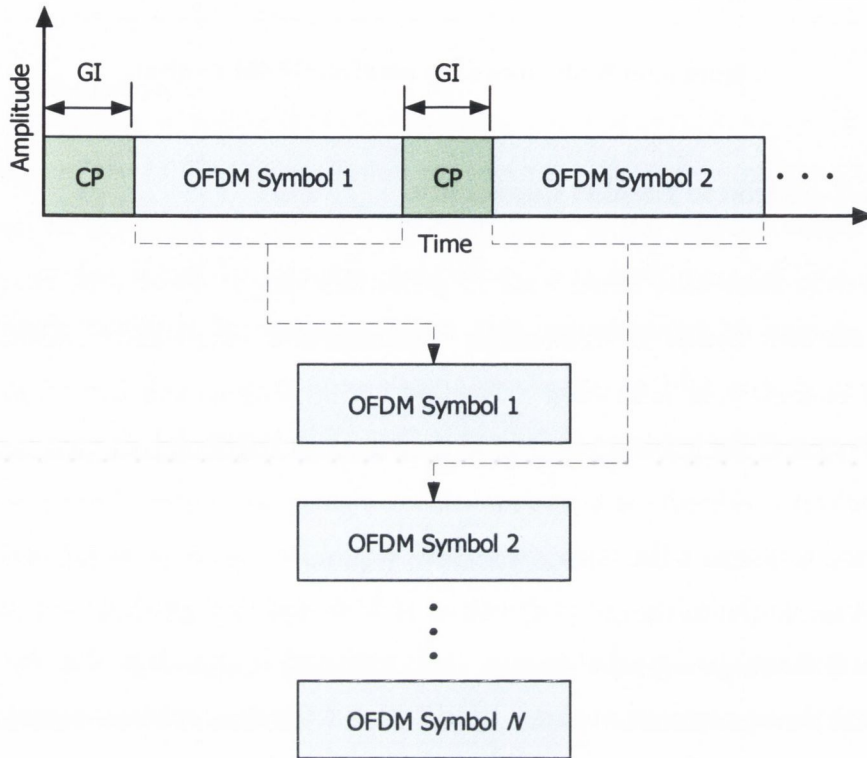


Figure 2.19: Conversion of a received OFDM frame into N parallel OFDM symbols.

2.6.3 Stage 9 – Fast Fourier Transform

Stage 9 is used to convert the real-valued received OFDM symbols back into a frequency domain representation of the signal where the value of each sub-carrier may be extracted. This stage performs the reverse of the original FFT process carried out in Stage 4 previously covered. The data symbols referenced by the sub-carrier indices that contain information data and not pilot symbols from the source may be extracted and then demodulated.

2.6.3.1 Sub-Carrier Extraction

The FFT block is a very important stage of the receiver as it enables the recovery of all the multiplexed sub-carriers without the need for multiple oscillators and filter combinations. The amplitude and phase values in the form of a complex valued number are resolved by

performing a FFT on each parallel block of received signals. Consider a discrete-time multi-path channel characterised by:

$$h(k) = \sum_{p=0}^{K-1} h_p \delta(k - \eta) \quad [2.9]$$

where $\delta(k)$ represents the dirac-delta function, $\{h_p\}$ is the set of complex path gains, $\{\eta\}$ is the set of path time delays considered in terms of multiples of OFDM samples, and K is the total number of propagation paths.

Assuming perfect sampling clock synchronisation, the received, down-converted and digitised signal samples may be expressed as:

$$r(k) = \exp\left(j \frac{2\pi k v}{N_{FFT}}\right) \sum_{p=0}^{K-1} h_p s(k - \eta) + n(k) \quad [2.10]$$

where $n(k)$ is a sample of complex-valued AWGN and v is the carrier spacing offset (or FFT array index increment).

Since perfect synchronisation is assumed and the guard interval has been already removed, the received signal sequence is therefore of the form $\{0, \dots, N_{FFT}-1\}$, and orthogonality has been maintained.

The n^{th} OFDM symbol may then be de-multiplexed enabling the $N_{FFT}/2$ sub-carrier values to be extracted from the FFT array, $Y(n)$, where $Y(n)$ may be expressed as:

$$Y(n) = \sum_{k=0}^{N_{FFT}-1} r(k) \exp\left\{-j 2\pi \frac{nk}{N_{FFT}}\right\} \quad [2.11]$$

2.6.4 Stage 10 – Channel Estimation and Equalisation

The received signals may have been affected by noise and fading channel effects prior to being processed by the receiver. To minimise the possibility of loss of data due to signal degradation, the effects of the channel must be estimated and then using this information, the receiver may then attempt to correct the signal data. The channel estimation and

equalisation stage, Stage 10, as shown in Figure 2.18, enables the receiver to compare a known signal sequence with the actual received sequence and then attempt to correct the remaining signal values. The known data sequence may be transmitted using the pilot carriers and if these carriers are transmitted along with each OFDM symbol or frame, the time-variant channel perturbations may be tracked and allow the correction factors to be updated upon reception of each OFDM symbol. Channel effects manifest themselves as variations in the phase and amplitude of the expected received signals. Channel estimation and equalisation is an important topic which will be dealt with in more depth later in this thesis.

2.6.5 Stage 11 – Parallel to Serial Conversion

This stage converted the parallel sequence of OFDM symbols into a serial sequence of OFDM symbols and is essentially the same operation as Stage 6 in Figure 2.2. Prior to the demodulation process, the number of OFDM symbols, N , in parallel must be converted to a serial sequence of OFDM symbols. This stage also results in the de-interleaving of the data symbols as the interleaving process described in Figure 2.7 is reversed.

2.6.6 Stage 12 – Symbol De-Mapping (Demodulation)

As shown in Figure 2.18, Stage 12, which is the symbol de-mapping, or demodulation stage is the final crucial stage in the OFDM receiver as it attempts to recover the original binary information that was transmitted from the received data symbols. Complex-valued signal points (or data symbols) are converted to binary data based on their phase and/or amplitude values. If the sub-carrier modulation scheme employed is BPSK, then the received complex-valued signals would be expected to occur in the vicinity of the two possible states as shown in the signal-space constellation diagram shown in Figure 2.3.

The actual received signals may deviate from the expected values due to residual channel estimation errors, discrepancies between the up-conversion and down-conversion RF oscillators, symbol and frame timing errors, and thermal noise due to the receiver RF front-end. A decision-metric algorithm may then be used to choose the binary value based on how close the actual received signal point is to the ideal constellation point for that particular binary digit. Figure 2.20 is a graphical example of eight arbitrary received data symbols representing the size of the byte (8 bits) used in the example described by Figure

2.7, normalised to unity and drawn on a constellation diagram. The modulation scheme used in this example is BPSK meaning that each symbol represents one binary value. Six of the eight data symbols are displaced from their expected constellation points to demonstrate that the symbols may be affected by noise or residual equalisation errors resulting in a phase and amplitude difference between the transmitted data symbol and the received data symbol value.

A decision-metric algorithm may be used to choose the constellation point closest to the actual received data symbol. In the example presented in Figure 2.20, using a hard-decision slicer, the received data symbols shown by the red signal points are resolved to the signal point $1 + j0$, and the signal points displayed in blue would be decided to correspond to the value $-1 + j0$. A binary one and a binary zero would then be recovered de-mapping the signal points $1 + j0$ and $-1 + j0$ respectively.

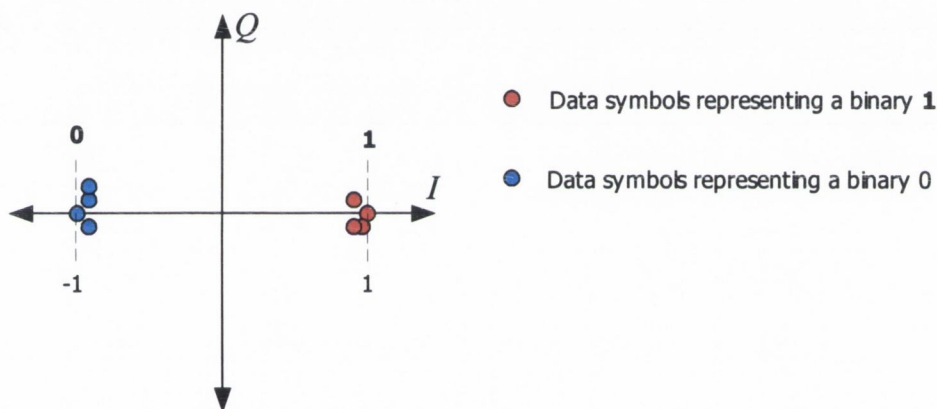


Figure 2.20: Example of received data symbols affected by noise and displaced from their expected constellation point values.

The output of this stage is a binary data stream, and in a perfect transceiver, it will be exactly the same as the original transmitted binary value sequence. The entire block diagram of the basic OFDM transceiver showing the transmitter and receiver previously examined is given in Figure 2.21.

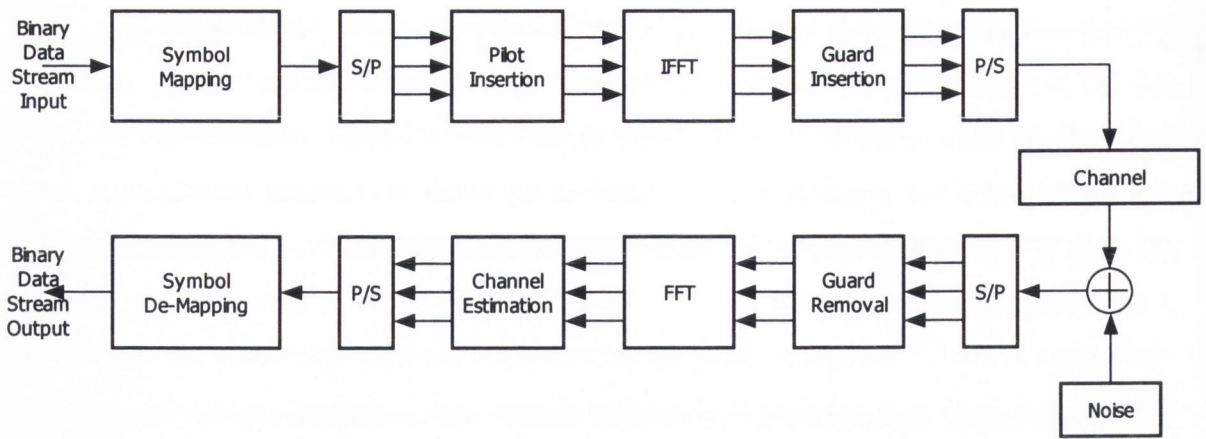


Figure 2.21: Basic block diagram of an OFDM transceiver.

Figure 2.21 shows the OFDM transmitter and receiver connected via a wireless communication channel affected by Additive White Gaussian Noise (AWGN) and multi-path fading effect. This is just a simple channel model as the transmitted signal may also be affected by Doppler fading effects, impulse and coloured (filtered AWGN) noise, in addition to AWGN and frequency-selective fading.

2.7 OFDM In A Challenging Wireless Environment

It has been shown that OFDM is a robust and spectral-efficient means of transferring information across a wireless communications channel. This section demonstrates how OFDM may be used to significantly improve the quality of wireless communications compared to traditional analogue communication techniques. A description of the main factors associated with transmissions in the frequency bands below 30 MHz is given. Many challenging wireless environments including variations of terrestrial and satellite-based communications links exist and transmitted signals may be affected by Rayleigh fading, Rician fading, Doppler fading and noise in addition to experiencing propagation models including tropospheric and ionospheric propagation. The following case study is one example of how OFDM, and a variant of OFDM called Coded-OFDM (COFDM), offer a better solution for narrow-band audio transmissions in the sub-30MHz spectrum segment. This is one example where ionospheric propagation can significantly influence the transmitted and received signals.

2.7.1 Fading and Interference

In the frequency bands below 30 MHz, the effects fading, noise and interference may significantly compromise the quality of received transmissions. COFDM is an OFDM transmission scheme where the input binary data are encoded using a convolutional code before the OFDM waveform is created. The received signals may then be decoded using a Viterbi decoder [Viterbi1967a] [Viterbi1967b]. Using these techniques, COFDM offers an increase in the robustness and error-resilience of the transmitted data.

In addition to multipath fading resulting from terrestrial wireless communications links due to artificial and natural RF reflecting bodies, the ionosphere also influences the propagation of wireless transmissions. OFDM can improve the quality of communications links in these channel conditions. The long, medium, and shortwave bands may be subject to considerable amounts of interference, fading and noise. There are two main ways in which a transmitted long, medium or shortwave radio signal may be propagated to a receiver over a wireless communications channel. The signal which travels directly to the receiver is known as the ground-wave signal. The second propagation method is the sky-wave mode, which means that the transmitted signal travels up to the electrically conducting layers in the atmosphere of the earth before being refracted towards the ground by these layers. The electrically conducting layers in the atmosphere are collectively known as the ionosphere and these layers are illustrated in Figure 2.22. The ionosphere is often characterised as comprising of a number of layers.

The main layers are named 'D', 'E' and 'F' layers, which are commonly used to denote the height at which the various ionisation effects due to the sun diminish in magnitude (i.e. ionisation effects are reduced as the distance to the earth decreases). The D-layer is the closest layer to the earth at approximately 50 km, while the F-layer may exist at a height of approximately 250 km from the earth. The ionospheric plasma of ionised particles may act as a refractive medium for radio waves if the density of the ionised particles is sufficient enough.

The height and density of these layers is dependent on the amount of charged particles emitted by the sun and travelling to the earth via the stream of rarefied plasma of protons and electrons emitted from the sun due to ejected matter from the sun. This plasma stream is called the solar wind and these particles may take a number of days to reach the earth's ionosphere. In addition, direct radiation from the sun in the form of impulsive bursts of Extreme Ultraviolet (EUV) and x-ray radiation result in enhancement of the ionisation level of the layers in the ionosphere. Solar flares generally occur near or in sunspots, which are areas of reduced temperature characterised by observable irregular dark patches on the sun. This sunspot cycle has an approximate duration of 11 years and dramatic propagation path elongation and increased duration may be observed at approximately the peak of the cycle when the number of sunspots is at its highest. The combined effect of solar radiation and the solar wind on the ionosphere is sky-waves incident on one or more of the ionospheric layers may either be absorbed or refracted towards the earth. The height of the ionised layer affects the distance between the transmitted signal and the point where the sky-wave returns to the earth. The signal at this point may be reflected back into the

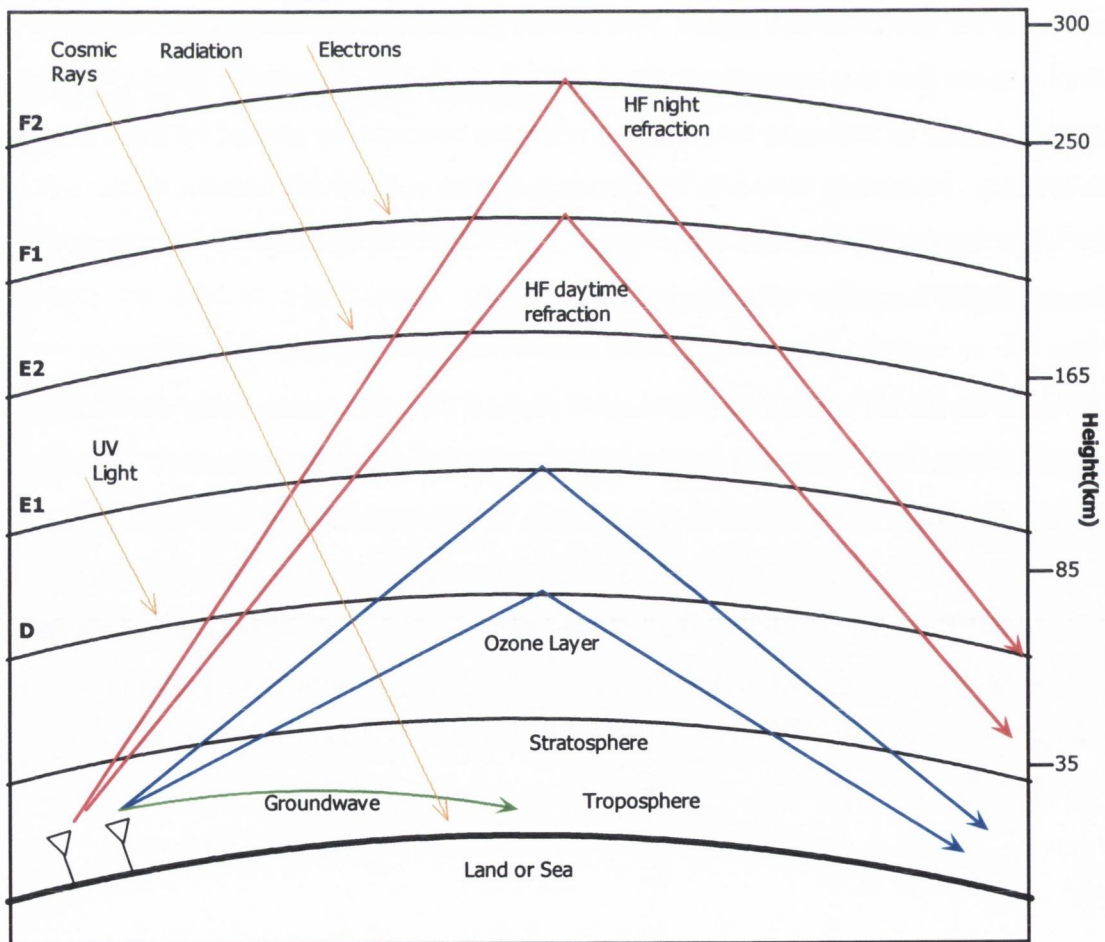


Figure 2.22: Layers of the ionosphere.

atmosphere by a land-mass or more effectively by large body of ionised particles such as seawater. This reflected signal may again be refracted towards the ground by one or more layers of varying height in the ionosphere. The effect of this phenomenon is that the sky-wave transmitted signal may travel many thousands of kilometres. When sky-wave refraction occurs, the transmitted signal may propagate significantly further than the ground-wave, which has a limited range and follows the curvature of the earth, from the same transmission source. The continuously varying height and density of these ionospheric layers means that a signal received via sky-wave propagation alone may suffer from fading as the received refracted signal component may vary in amplitude and even the location where the refracted signal returns to the earth again.

Fading is an especially dominant factor in the long, medium, and shortwave bands mainly due to the effects of the troposphere and ionosphere on a transmitted signal. The ground-wave propagated along the curvature of the earth and the sky-wave travels up into the atmosphere. As a result of sky-wave propagation, many different multi-path signal components may arrive at the receiver, using several different propagation paths, and the resulting signal may comprise many signal contributions of different amplitudes, time lags, and possibly subject to frequency-selective fading. If the receiver is in the range of the ground-wave signal component, then this signal may arrive at the receiver slightly before the sky-wave signal components thus compounding the problem of multi-path fading. The ionosphere may also affect the polarisation of one or more of the signals resulting in phase differences between the received signal components therefore a rugged and efficient modulation/access scheme is required in order to be able to offer reliable reception of broadcast transmissions. As a result of the possibly very long propagation paths that transmitted signals may encounter, interference is commonplace as co-channel transmissions may propagate far and interfere with other. Lightning results in impulsive noise in these frequency bands in addition to noise sources from overhead power lines and RF welding devices, amongst many others.

The characteristics of signal propagation due to the refractive layers in the ionosphere are dependent on the transmission frequency. Long and medium wave transmissions generally propagate further during the hours of darkness and higher-frequency signals may propagate further during the day. These characteristics are due to the nature of the different layers in the ionosphere, which may vary in height and density according to the time of day [Malaga1979]. This is illustrated in Figure 2.22, where the layer associated for

propagation of HF signals during the day is generally the D-layer and during the night time, the much higher F-layers are the predominant refractive medium. This effect of these is especially apparent with long and medium-wave transmission which propagate significantly further during the hours of darkness. It is important to note that the refractive properties of the layers in the ionosphere are frequency dependent and there exists a Maximum Usable Frequency (MUF), which is exceeded, means that the incident signal is not refracted towards the ground but continues travelling towards space. The MUF is also dependent in the time of day and level of solar activity [PESWG1991].

OFDM/COFDM systems can increase their robustness against the significant fading effects by choosing different types of guard-intervals between the COFDM symbols in addition to interleaving the transmitted information over a number of OFDM symbols. The coherence bandwidth of the channel is dependent on the type of signal propagation that may be experienced. A long guard interval helps reduce ISI that may result from the many different multi-path signals occurring with long-distance sky-wave propagation. A shorter guard-interval is suitable for local or regional coverage using ground-wave propagation as the coherence bandwidth is expected to be reduced.

A large number of evenly spaced sub-carriers may be used to generate the COFDM symbol. The modulation schemes employed may be Quadrature Amplitude Modulation 4-QAM, 16-QAM or 64-QAM. It is therefore possible to use modulation schemes with a high bit-to-symbol ratio in order to maximise the amount of information transmitted through the wireless channel. Guard intervals are appended to the transmitted symbols in order to ensure a high degree of resistance to inter-symbol interference due to multi-path fading effects caused by sky-wave propagation. The encoded information is conveyed using a similar process of sub-carrier insertion and multiplexing to form a sequence of COFDM symbols.

The high-data rate and spectral-efficient nature of OFDM/COFDM radio systems enables digital audio in the form of multi-lingual speech and music to be broadcast. Images, news and other messages may also be transmitted to a very wide audience. Reception of these transmissions is more reliable than normal analogue techniques as frequency-diversity helps to combat frequency-selective fading and interference, inter-leaved data techniques help prevent loss of information from flat-fading effects and noise, and convolutional coding offers a level of error-correction to create a rugged and relatively error-resistant

broadcasting system. The entire bandwidth of the channel may be used for data transmission or divided up into sub-bands, with each sub-band capable of carrying different sources of information or data types. OFDM/COFDM radio systems are capable of providing near-FM quality audio and integrating data and text, a concept not possible with traditional analogue transmission schemes present on the long, medium and shortwave bands.

2.8 Conclusion

The chapter introduced a multi-carrier modulation technique called OFDM. The basic principles of how an OFDM modulator and demodulator may be created were presented. It has also been discussed that OFDM is an efficient and robust method of transmitting information over wireless communications channels affected by noise, fading effects and interference than a single-carrier transmission system. Due to the high data-rate and robust nature of OFDM, it has become a very popular wireless transmission technology.

3

CURRENT OFDM IMPLEMENTATIONS

3.1 Introduction

As stated in the previous chapter, OFDM has become a very popular means for high-rate digital communications and is steadily gaining even more popularity as a transmission technology. OFDM offers a high degree of robustness against noisy and fading channel conditions thanks to its frequency diversity, block interleaving techniques and the extra robustness due to Coded-OFDM (COFDM). This section describes some radio systems that use OFDM for broadcasting sounds and images, in addition to enabling high-speed data links between fixed and/or mobile transceivers. An in-depth analysis of the entire transceiver structure and protocols used may be found in the referenced works for each technology. The descriptions of the transceiver technologies given in this section extend to an overview of the particular applications implemented and the methods used to convey information across a wireless communications channel using OFDM.

3.2 IEEE 802.11

One of the most common uses of OFDM is to facilitate high-speed Wireless Local Area Networks (WLAN). One of the most popular WLAN standards in use today is the Institute of Electrical and Electronics Engineers, Inc. (IEEE) 802.11 standard [IEEE802.11] and its variations, which will be discussed in later sections of this chapter. In the context of this thesis, the main focus of the treatment of WLANs is in regards to the methods used by the devices to access the wireless channel, and with special attention to the schemes that employ OFDM. 802.11 actually refers to a family of specifications developed by the IEEE for WLAN technology which was accepted by the IEEE in 1997. IEEE802.11 specifies an over-the-air (OTA) interface between a wireless device (client) and a base station. This standard also specifies an OTA interface for use between two wireless devices.

These products are designed as a replacement for the typical wired-Ethernet connections but unlike wired-Ethernets, wireless local area networks are subject to propagation, noise and other time-variant effects synonymous with wireless communication channels. A WLAN protocol stack or simply 'stack', defines the way information may be passed to/from a source or destination and converted to/from binary sequences and electromagnetic wave representations. This stack comprises several layers of functionality and signal processing and each layer is responsible for one or more particular signal processing/management tasks with the protocol stack. High-level layers deal with the binary input/output procedures, in effect, the application which is 'visible' to the user. A Medium Access Control (MAC) layer may be close to the lowest layer in the stack and this controls the access to the wireless channel using a protocol specific to the intended application. The specific modulation schemes/channel access schemes and bands of interest are determined by the bottom layer of the stack known as the Physical Layer (PHY). WLANs may use different methods of transmitting and receiving waveforms using a wireless channel hence a set of WLAN standards was agreed upon by the IEEE to define different physical layers that may be used. Two of the IEEE wireless communication standards used for WLAN devices involve the use of OFDM.

The underlying idea of the 802.11 implementation is to abstract the main signal-processing application (i.e. the protocol stack) away from the RF physical front-end. This means that potentially different variations of the 802.11 standard may be implemented using the exact same RF front-end. This enables sub-layers within the physical layer entity to control the method in which the information is transmitted. The result is that the operation of the physical layer is not tied to the actual 802.11 standard that is implemented. This is therefore of particular relevance to the work presented in this thesis as one of the main objectives is maintain independence between the physical hardware of the RF front-end and the actual radio implementation. The radio implementation may change but the RF front-end may not. More details of this concept will be presented later in this thesis.

Each physical layer used for all the IEEE WLAN standards may consist of two protocol functions, controlled by a physical layer management entity, which are shown in block form in Figure 3.1 and described as follows:

1. A Physical Medium Dependent (PMD) system, which is the radio front-end hardware.

This defines the characteristics of, and method of transmitting and receiving data through, a wireless medium (WM) between two or more WLAN transceivers. In other words, the PMD sub-layer provides a transmission interface used to send and receive data between two or more WLAN devices. This PMD is shown as the lowest sub-layer in the physical layer illustrated in Figure 3.1.

- 2. A Physical Layer Convergence Function (PLCP), which acts as an intermediate stage between the physical radio front-end and the wireless service required.

This adapts the capabilities of the radio front-end hardware to the required wireless service. To allow the IEEE 802.11 MAC to operate with minimum dependence on the PMD sub-layer, the PLCP simplifies provision of a physical layer service interface to the IEEE 802.11 MAC services. In effect, this function defines a method of mapping the IEEE 802.11 medium access control (MAC) sub-layer Protocol Data Units (MPDUs) into a framing format suitable for sending and receiving user data and management information between two or more wireless stations using the associated PMD system. The PLCP sub-layer is shown as the top sub-layer in the physical layer block diagram described in Figure 3.1. A PLCP preamble provides a period of time for several receiver functions. These functions include antenna diversity and clock and data recovery.

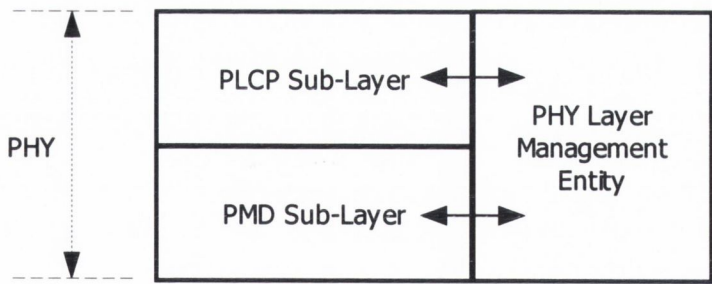


Figure 3.1: PMD layer reference model.

3.2.1 The 802.11b Standard

The first 802.11 standard of the mid-1990s resulted in the 802.11b standard [IEEE802.11b], which was approved by the IEEE in 1999. The physical layer for this particular standard does not involve the use of OFDM but serves as a useful introduction to the two OFDM-based standards, IEEE802.11a and IEEE802.11g, which both use OFDM.

These two variations of the 802.11 standard will be described in later sections. The standard dictated by the 802.11b specification stated that chip sets would use a modulation scheme known as Complementary Code Keying (CCK) to transmit data signals at 11 Mbps through a portion of the unlicensed Industrial, Scientific, and Medical (ISM) 2.4GHz band (between 2.402 GHz and 2.4835 GHz in Europe and North America). The maximum Effective Isotropic Radiated Power (EIRP) is limited to 100mW. This 802.11b standard, which uses Frequency Hopping Spread Spectrum (FHSS) or Direct Sequence Spread Spectrum (DSSS), has been largely superseded by a new generation of wireless LAN devices that enabled wire-free Ethernet connections but still at a much more reduced data transfer rate than wired-Ethernet connections.

3.2.1.1 The 802.11a Standard

An extension of the IEEE 802.11 standard is 802.11a [IEEE802.11a], which uses OFDM and operates in the 5 GHz band. The OFDM transceiver system used in this standard is capable of transferring data at 6, 9, 12, 18, 24, 36, 48, and 54 Mbps. 52 sub-carriers are used and the array of transfer rates possible is because the data symbols associated with these sub-carriers may use binary or quadrature phase shift keying (BPSK/QPSK), 16-quadrature amplitude modulation (QAM), or 64-QAM. The input binary data are encoded using a convolutional coding scheme with coding rates of $1/2$, $2/3$ and $3/4$. Forward Error Correction is used at the receiver to correct any errors that may have occurred in the received and demodulated sequence of binary values. An outline of the transmitter and receiver used in the 802.11a physical layer is shown in Figure 3.2. The main signal-processing stages in the signal chain shown in this diagram are similar to the operation of the OFDM modulator and demodulator diagram previously described in Chapter 2.

The transmitter block diagram shown in Figure 3.2 includes the convolutional-coding Forward Error Correction (FEC) functional block, which precedes the bit-interleaving and mapping stage. The up-conversion stage is also included in the form of a RF oscillator mixer stage prior to the High Power Amplifier (HPA), which amplifies the signal to the maximum allowed transmit power, and transmit antenna. This receiver block diagram also includes the down-conversion and the stage which converts the real-valued received and down-converted signal into its In-phase (I) and Quadrature (Q) signal representation. Also included in this diagram is the Automatic Gain Control (AGC) used to attenuate very strong received signals and also to amplify low-powered intercepted signals. Apart from

the stages preceding the IQ Modulator in the transmitter and the stages preceding the de-mapping/de-interleaving stage in the receiver, the signal waveform is an analogue signal due to the continuous-time RF oscillators used for up/down-conversion.

A table of the major parameters specified in the 802.11a standard is shown in Table 3.1. The total bandwidth of the pass-band signal is 16.6 MHz using 52 sub-carriers. The duration of the useful part of each OFDM symbol is $3.2\mu S$. A guard interval of duration $0.8\mu S$ is pre-pended to the useful part of the OFDM symbol to form the transmitted OFDM symbol, which has a total duration of $4\mu S$. The highest data rate (54 Mb/sec) is achieved using convolutionally-encoded binary data modulated using 64-QAM. Certain bits from the encoded bit sequence are removed as a means of increasing the data rate. The Viterbi decoder [Viterbi1967a] at the receiver is then used to recover these lost bits. This bit-removal process is called ‘puncturing’ and the coding algorithm is known as a ‘punctured code’.

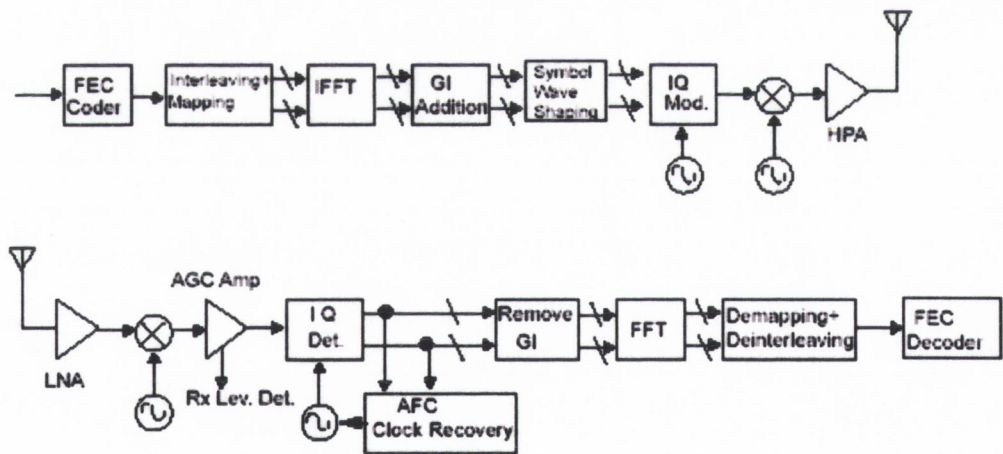


Figure 3.2: Transmitter and receiver block diagram used for the 802.11a PHY (taken from the IEEE802.11a specification [IEEE802.11a]).

Table 3.1: Table of the major parameters of the OFDM physical layer for 802.11a (taken from the IEEE802.11a specification [IEEE802.11a]).

Information data rate	6, 9, 12, 18, 24, 36, 48 and 54 Mbit/s (6, 12 and 24 Mbit/s are mandatory)
Modulation	BPSK OFDM QPSK OFDM 16-QAM OFDM 64-QAM OFDM
Error correcting code	K = 7 (64 states) convolutional code
Coding rate	1/2, 2/3, 3/4
Number of subcarriers	52
OFDM symbol duration	4.0 μ s
Guard interval	0.8 μ s ³ (T_{GI})
Occupied bandwidth	16.6 MHz

3.2.1.2 The 802.11g Standard

This standard relates to a similar OFDM-based transceiver as a WLAN using 802.11a except that 802.11g [IEEE802.11g] operates in the 2.4 GHz band. This standard also includes the same channel access and modulation scheme capabilities used for 802.11b. Therefore, 802.11g is backwards compatible with the 802.11b standard. The implications of this mean that existing 802.11b infrastructure may be upgraded to 802.11g, without the requirement for extra hardware (if 802.11g is already the chosen device used for 802.11b implementations) resulting in faster data rates. Figure 3.3 shows the high-level block diagram of the transmitter used in an 802.11g implementation. The OFDM signal processing kernel and the 802.11b processing kernel both required access to the Digital to Analogue Converter (DAC) and the RF front-end. A soft-switch, which is a software means of selecting one of many signal paths, is used to switch between the two different kernels when required. The average transmission power from both the OFDM and 802.11b kernels must be equal however. This is to ensure that the operation of rest of the stages following the soft-switch remains within the parameters

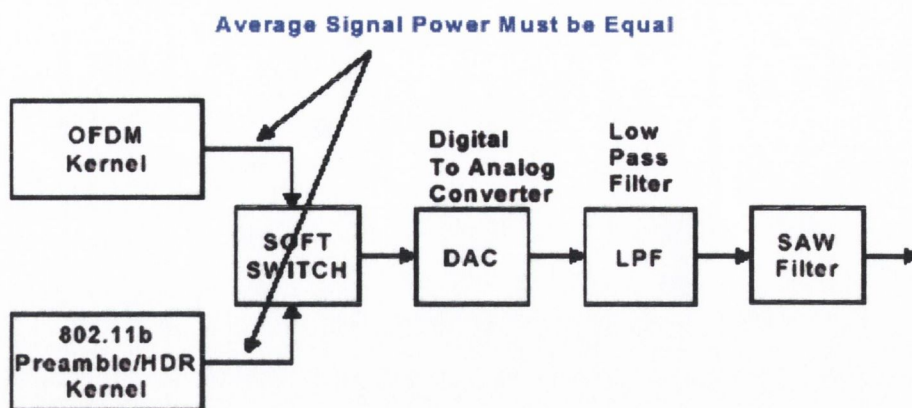


Figure 3.3: OFDM/802.11b transmitter high-level block diagram (taken from the IEEE802.11g specification [IEEE802.11g]).

3.2.2 IEEE 802.11 Medium Access Control (MAC)

This section examines the main aspects of the common radio management and data structures that are used within the 802.11a/b/g and 802.16 (which is described in the next section) WLAN standards. These are of particular importance to this thesis as they are good examples of how channel bandwidth is wasted from overheads used to coordinate the receiver. These overheads are necessary for the 802.11 WLAN implementations but a better solution involving a receiver with enhanced functionality may mean that more of the channel bandwidth can be devoted to transferring valid information from the source(s).

The main objective of a wireless LAN is to transport data as quickly and as reliably as possible between two or more transceivers. 802.11 defines a data frame type that carries packets from higher layers such as the application. The body of this frame may carry audio, images, web pages, or control information. Extra information such as the source address and destination address of the information, in addition to other protocol information are appended to this data frame in the form of a control field.

The Medium Access Control (MAC) used in 802.11 is intended to be independent of the physical layer (PHY) therefore sub-layers and procedures are used to simplify the PHY interface instead of implementing a fixed, physical layer-dependent method of accessing the physical layer. 802.11 uses Physical Layer Convergence Procedure (PLCP) and Physical Medium Dependent (PMD) sub-layers. These are terms that the standard uses to divide the major functions that occur within the physical layer. The PLCP prepares 802.11 data packets (frames) called physical sub-layer service data units (PSDU) for transmission

and directs the PMD to actually transmit the signals. The PLCP also directs the PMD to start receiving signals, and switch channels if required. The PLCP takes each PSDU that requires to be transmitted and forms what the 802.11 standard refers to as a PLCP Protocol Data Unit (PPDU).

A PPDU not only contains the OFDM symbols that contain the binary data from the source, but also information regarding the modulation scheme that is required, which is specified by a code that specifies the required data rate and coding scheme used. The structure of the PPDU frame and the PLCP header is shown in Figure 3.4. This PPDU frame contains a preamble used at the receiver to enable it to acquire the signal. Timing and coarse frequency synchronization in addition to enabling the Automatic Gain Control (AGC) loop to establish a lock on the signal is also facilitated using this preamble. It comprises ten repetitions of a training sequence, which is a sequence of length $16\mu\text{s}$ consisting of 12 sub-carriers modulated using a pre-determined data symbol sequence. Following the preamble is the PLCP header, which is one OFDM symbol containing information relating to the type of coding and modulation scheme that is in use. This header is convolutionally coded using a rate 1/2 code (one un-coded bit in results in two encoded bits out) and modulated using BPSK. The data field in the PPDU contains the original PSDU in addition to extra bits used to synchronise and clear receiver processing functions.

The received transmissions from the 802.11 device is extracted and re-structured in a PPDU format. The receiver then proceeds to ascertain whether a valid PPDU has actually been captured. The receive operation as specified in the 802.11a specification, waits until the preamble (training sequence) is received. A Received Signal Strength Indicator (RSSI), which registers a signal power increase associated with this training sequence, notifies the PLCP upon reception of this preamble. The physical layer then searches for the PLCP header, as shown in Figure 3.4, contained in the received PPDU in order to establish how long the received data stream is, the modulation scheme used, and the coding rate used. A correctly received signal (i.e. error-free) is indicated by the parity bit. The PSDU may then be extracted and decoded using a Viterbi decoder.

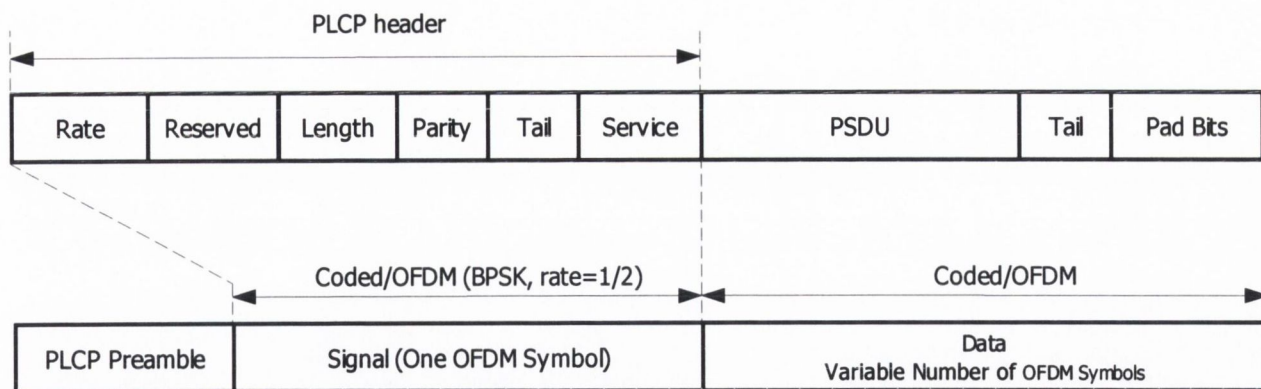


Figure 3.4: PPDU frame format.

3.2.3 Proposed 802.11n Standard

The next OFDM-based wireless standard being discussed by the IEEE is the 802.11n [IEEE802.11n] specification proposal, which is an example of how OFDM will be the key to achieving significantly faster data rates than current commercial offerings. It is expected that this specification will combine OFDM and Multiple Input Multiple Output (MIMO) technology. A MIMO system, which is a space-diversity transmission scheme, uses two or more antennas for transmission and reception creating two or more independent transmission paths. Information may then be transmitted in parallel using these independent transmission paths. An 802.11n implementation using an array of four antennas for transmission and reception of an OFDM signal, using a modulation scheme with a high bit to symbol ratio such as 64-QAM, means that data rates up to 540 Mbps may be achieved.

3.3 The 802.16 Standard (WiMAX)

Designed for use in Metropolitan Area Networks (MANs) the 802.16a [IEEE802.16a][IEEE802.16b][IEEE802.16c] specification includes OFDM as one of transmission technologies used for high-rate data transfer. The original 802.16 standard was published in April 2002 and specified fixed point-to-multipoint broadband wireless systems operating in licensed spectrum in the 10 to 66 GHz range to deliver up to 70 Mbps at distances up to approximately 50 km. A subsequent amendment to the original standard, resulting in the 802.16a specification was approved in January 2003. This amendment extended the capabilities of the 802.16 standard to include non-line-of-sight operation in the 2 to 11 GHz frequency allocation. OFDM is mandatory for use within the licence-

exempt portions of the 2 to 11 GHz band, and due to the possible non-line-of-sight operation, occurrences of multi-path fading are highly possible. OFDM is therefore a suitable candidate for this scenario. Officially called the WirelessMAN™ specification, or WiMAX, the 802.16 standards are designed to enable multimedia application wireless connectivity using an alternative to fibre optic and cable network access and their associated high infrastructural costs.

3.4 Digital Audio Broadcasting (DAB)

Another technology that exploits the capabilities of OFDM and is worth discussing is the Eureka-147 DAB System. In 1994, THE ETSI (European Telecommunications Standards Institute) issued ETS 300 401 [ETS300401DAB], a European DAB standard which was proposed based on earlier work on the Eureka-147 DAB system, which also uses OFDM. DAB represents a dramatic advance in radio technology since the introduction of FM stereo radio.

This was designed to be a robust broadcasting system, and thanks to the use of OFDM, it was also designed to be a highly spectrally efficient sound and data broadcasting system. Creating a power-efficient means of delivering, and receiving, high-quality sound (CD quality) was also another main goal for this project. DAB uses digital audio compression techniques (MPEG 1 Audio Layer II and MPEG 2 Audio Layer II), which encodes the information before OFDM is used to convey this data at a high data-rate to a mass audience. DAB was also designed to be a multi-service digital broadcasting system where several sources of information such as news, weather, time and date may be multiplexed for transmission using a single channel bandwidth. The audience has to be able to receive the DAB signal using mobile, portable and fixed receivers with only a simple, non-directional antenna

DAB is versatile in regards to its operating frequencies and implementation. It was envisaged that it may operate at any frequency from 30 MHz to 3 GHz for mobile reception (higher for reception by a fixed receiver). In addition, the network infrastructure used to deploy DAB was aimed at terrestrial, satellite, hybrid (satellite with complementary terrestrial) and cable broadcast networks. Extra information sources such as news, weather, control data may be multiplexed along with the main service material. The advantages include interference-free sound, additional stations and services, in addition to

an easy to use radio, capable of selecting or browsing stations by name instead of by frequency. Differential QPSK ($\pi/4$ -shift DQPSK) is used for the ETSI DAB standard. This modulation relies on the phase difference between successive data symbols to represent the modulated information. The binary information is interleaved in time and frequency, in addition to a Rate-Compatible Convolutional Code (RCPC) to increase the robustness of the DAB signal when transmitted over a channel subject to fast fading, noise, and interference.

Two data transport mechanisms are defined in the DAB standard. One of these mechanisms is known as the Fast Information Channel (FIC), which contains control information that is used to disseminate the information carried by the MSC as well as information of the types of services that are conveyed by the MSC. This FIC also has the provision for paging, Traffic-Message Channel (TMC), and an Emergency Warning System (EWS).

The second data transport mechanism is the Main Service Channel (MSC). The MSC comprises interleaved data frames referred to as Common Interleaved Frames (CIF). A CIF carries 55,296 bits, which is transmitted every 24ms, and each CIF is addressable in 864 64-bit sections called Capacity Units (CU). The basic transport unit, or sub-channel, of the MSC is created by grouping integral numbers of these capacity units. Each sub-channel may be assigned to a particular service such as audio, news, ticker-tape stock market news or even advertisements. Several of these sub-channels are then multiplexed to form the complete MSC.

3.4.1 Audio Encoding

Audio Encoding is an interesting part of digital audio broadcasting as it enables much higher quality audio to be transmitted using a digital modulation scheme such as OFDM. Traditional analogue transmission schemes transmit analogue audio, which is often band-pass filtered to reduce the bandwidth (and therefore reducing the quality) of the sound. An audio encoder means that a higher-fidelity sound may be transmitted and received using a lower bit rate.

The DAB system uses MPEG Audio Layer II for encoding audio streams. If the sampling rate is 48 kHz then the ISO/IEC 11172-3 standard is used and for 24 kHz sampling frequency, the ISO/IEC 13818-3 standard may be employed [ISO/IEC94][ISO/IEC93].

The input Pulse Coded Modulation format (PCM) audio signal, sampled at either 48 kHz or 24 kHz is encoded and the resulting compressed audio bit stream is produced. The encoder may produce compressed audio streams with bit-rates ranging from 8 kbps to 384 kbps. Four audio modes may be used for audio encoding:

- Single channel (i.e. mono) mode.
- Dual channel (i.e. two mono channels) mode.
- Stereo mode.
- Joint stereo mode.

3.4.2 Transmission Modes and DAB Frame Structure

A DAB transmission frame, as described in Figure 3.5, consists of a sequence of synchronisation channel symbols, Fast Information Channel (FIC) symbols and Main Service Channel (MSC) symbols. The synchronisation channel symbols comprise a null symbol used to identify the start of a frame, and a phase reference symbol, which is used as a starting phase value for the DQPSK demodulation. These synchronisation symbols are used internally within the transmission system for demodulator functions such as transmission frame synchronisation, Automatic Frequency Control (AFC), channel state estimation, and transmitter identification. The MSC symbols contain the multiplexed services, which may be extracted correctly by using the information carried by the FIC symbols, which in turn are obtained when the receiver identifies the start of the DAB frame using the synchronisation channel symbols. The duration of this frame, specified in seconds, is denoted as T_f .

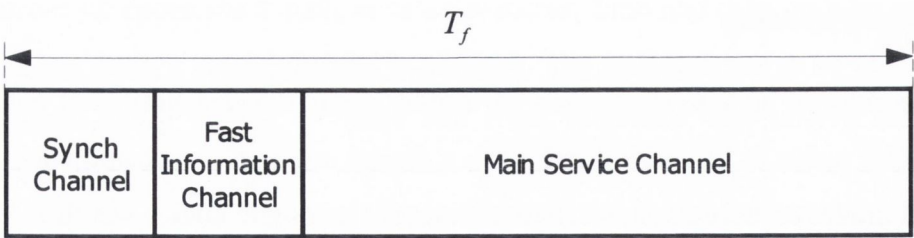


Figure 3.5: DAB frame structure showing the arrangement of the synchronisation (sync) channel, fast information channel and the main service channel.

DAB uses four different transmission modes, denoted modes I, II, III and IV. These are country-specific and required for worldwide DAB reception. Mode I is for Band III (DAB frequency band 174 – 240 MHz) and terrestrial transmission, Mode II is for the higher frequency L-Band (DAB frequency band 1452 – 1492 MHz) used for both terrestrial and satellite transmissions, Mode III is used for terrestrial and satellite DAB transmission using

frequencies below 3 GHz. Mode IV is designed for L-Band, terrestrial and satellite distribution. The duration of each DAB frame used for mode I is 96ms, modes II and III use frame durations of 24ms, and the duration of a DAB frame for mode IV is 48ms. A DAB receiver must be capable of dealing with all four transmission modes.

A block diagram of a typical DAB modulator/creation process is shown in Figure 3.6. This example illustrates the DAB principle from the service information input stage through to the production of a sequence of COFDM symbols. Three services are multiplexed in this example, including the creation of the CIF and FIBs which are not shown in this diagram in order to maintain the clarity of the description. The resulting multiplexed binary sequence is partitioned into blocks (or shorter sequences of binary values). The next stage involves converting the binary values to signal points using a QPSK constellation. Every two-bit sequence is transformed into a single complex-valued QPSK data symbol. The frequency-interleaving stage simply re-orders the data symbols in according to a permutation dictated by the mode of transmission. For each of the four transmission modes, there is a specific integer set that determines the order of the data symbols. The phase reference symbol, used as the starting phase point for finding the phase difference between successive DQPSK symbols, and the interleaved QPSK data symbols are converted to a sequence of DPSK symbols. This modulated information is therefore represented by the difference in phase between two symbols and not the actual phase value itself. This DQPSK sequence, along with a synchronisation sequence comprising null data symbols, and the phase reference symbol itself are then multiplexed to form the final data symbol sequence. This final sequence comprises the data frame as illustrated in Figure 3.5. Finally, the data frames are then converted to OFDM symbols using the OFDM symbol generator processing block. After these waveforms have been amplified and frequency-translated to the band of interest, the DAB signal is transmitted.

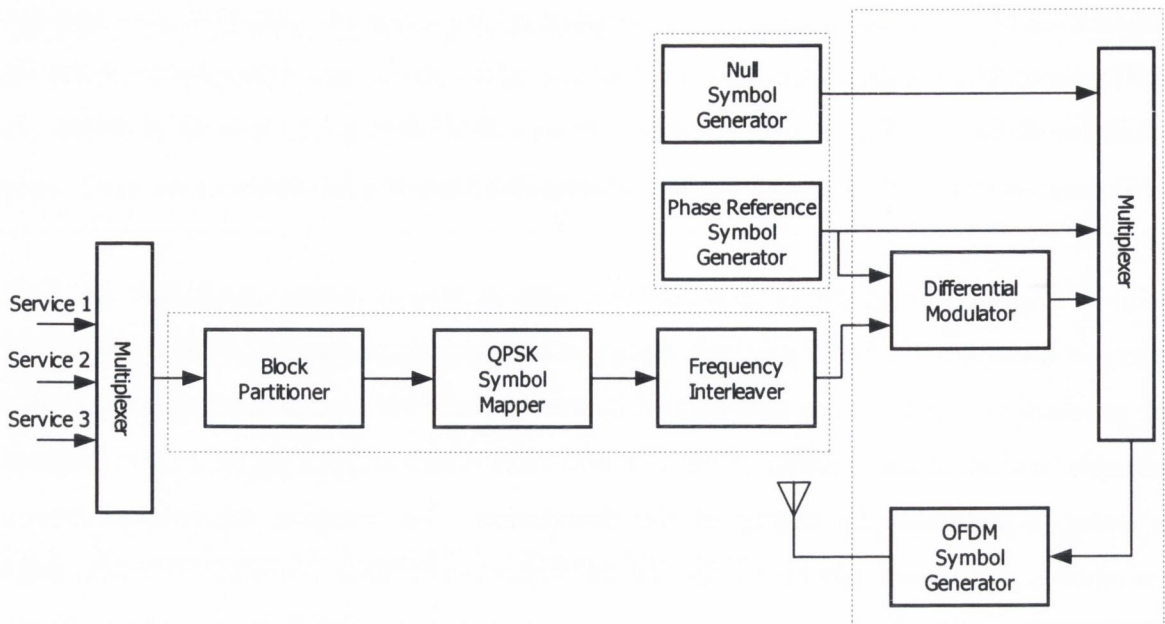


Figure 3.6: Block diagram of a DAB modulator.

From June 2004, a number of UK national DAB services in the form of a multiplexed DAB signal were available from the British Broadcasting Corporation (BBC) including BBC Radio 1, 2, 3 and 4, BBC World Service, Asian Network, BBC Radio Five Live Sports Extra, and data services like BBC Vision Radio and an Electronic Programme Guide (EPG). Music services are typically broadcast at 128 kbps stereo within the United Kingdom, and mono speech services are encoded at 80 kbps. DAB services within Europe are mainly concentrating on transmitting national services already available on the VHF FM frequency bands in the form of DAB ensembles [WorldDAB]. This example shows that OFDM is helping to revitalise broadcast radio services by being facilitating the delivery of much higher quality audio and a large variety of programme material to wide audience.

3.5 Digital Video Broadcasting-Terrestrial (DVB-T)

Digital Video Broadcasting (DVB) [DE/JTC-DVB-8] is a third example of an OFDM-based multi-media broadcasting technology that is rapidly gaining popularity as a possibly eventual replacement for existing terrestrial analogue audio and video broadcasting systems. Founded in September 1993, the Digital Video Broadcasting (DVB) project was developed by consortium of public and private sector organizations in the television industry. Its aim was to establish a framework for digital television services using MPEG layer 2 coding for video compression and MPEG layer 3 coding for the audio streams. In

addition to the terrestrial DVB (DVB-T) service, satellite DVB (DVB-S), DVB over cable networks (DVB-C) standards are also being developed but for the purposes of this chapter, only DVB-T is of relevance as it uses COFDM as the transmission technology.

In a similar fashion to DAB, the audio and video streams are multiplexed and partitioned into blocks of data. Each data block is then packaged with service information used to identify the contents of the DVB frame and to aid the conversion of the received information back to audio and video streams. Convolutional coding is used in DVB-T in addition to a Viterbi decoder at the receiver. As for the DAB standard, convolutional coding is used to reduce the bit-error rate of the received sequence of binary values and to increase the robustness of the broadcasting mechanism.

High quality video and audio broadcasting demands a very high transmission data rate. In order to facilitate this data rate, up to 6817 sub-carriers are used to generate the sequence of OFDM symbols. DVB may use one of two modes. The first mode is denoted the '2k mode', which uses 1705 sub-carriers and the second case is the '8k mode', which uses 6817 sub-carriers. For single transmitter operation, the '2k mode' is suitable for single transmitter operation and for small single frequency networks with limited transmitter distances. The '8k mode' is mainly used for both single transmitter operation, and for small and large SFN networks. The sub-carriers include pilot symbols used for frame synchronisation, frequency synchronisation and equalisation. One DVB transmission frame contains 68 OFDM symbols and one super-frame contains four transmission frames.

All of the interleaved symbols present on the data carriers in one OFDM transmission frame are modulated using QPSK, 16-QAM, or 64-QAM. DVB-T is currently being used to transmit digital television channels such as 'Freeview', which is jointly owned by the BBC, Crown Castle International and British Sky Broadcasting. It broadcasts the existing free-to-air British-based analogue channels (BBC1, BBC2, ITV, Channel 4 and Channel 5) in addition to several other channels provided by the BBC, ITV companies, Sky, Crown Castle, and independent companies. It is envisaged that OFDM will feature even more predominantly as an attractive and efficient means of delivering high-quality audio and video services to a mass audience.

3.6 Digital Radio Mondiale (DRM)

In Chapter 2, Section 2.7, a description of how OFDM may be used to increase the quality of received broadcast transmissions in the noise, fading and interference-prone frequency bands below 30 MHz was presented. Digital Radio Mondiale (DRM) [RES/JTC-DRM-07] is a broadcast technology that uses OFDM to provide this service. DRM is a universal non-proprietary digital AM radio system that uses COFDM to broadcast near-FM quality sound and data in the long, medium, and shortwave bands. DRM is an excellent example of an OFDM system that is designed for higher quality transmission of sound, images and data in this frequency range.

With the introduction of high-quality digital audio broadcasting using satellites, FM, and radio broadcasting using the internet, the attraction of the long, medium, and shortwave bands to broadcasters and listeners alike is diminishing. The reason for this is that the frequency bands below 30 MHz were traditionally associated with variable-quality broadcast reception, in addition to the fact that these frequency bands are particularly susceptible to interference, noise and fading, as was discussed in the previous chapter. In an attempt to radically improve the quality and appeal of the broadcast band segments below 30 MHz, a digital audio scheme using OFDM was designed for these bands. DRM is digital audio broadcasting but uses the same bandwidth as the existing broadcast analogue AM radio stations but by exploiting the ability of OFDM to convey digital data using many different frequencies simultaneously. As a result, a much higher quality sound is achievable in addition to extra information such as programme listings, images and news messages. The DRM signal was designed to fit in with the existing AM broadcast plan therefore the total bandwidth of the DRM signal is 9 kHz for medium wave transmissions in Europe, 10 kHz medium wave transmissions in North America or up to 20 kHz for transmission in the shortwave bands. Typically, a listener may have to retune their short-wave receiver possibly several times during the course of a day as the channel conditions and signal propagation characteristics of these frequency bands may vary dramatically over the course of a day. DRM enables extra information to be transmitted along with the main programming substance such as alternative frequencies that the receiver may retune to automatically if desired.

An international consortium consisting of manufacturers, broadcasters, research groups, regulatory bodies and broadcasting unions, originally formed in 1998 from meeting of a

representatives from Radio France Internationale, TéléDiffusion de France, Deutsche Welle, Voice of America, and Thomcast., joined forces to create a universal, digital audio system, called Digital Radio Mondiale. A consensus emerged from this initial meeting that unless a means of increasing the value of the broadcast transmissions in the bands below 30 MHz by improving the sound quality and reception reliability, the outlook for these bands would be bleak. In a response to this challenge, the DRM system, using COFDM was conceived and this project is being led by VT Merlin Communications [VTMerlin]. This digital audio radio system was designed for use in the long, medium and shortwave bands in an attempt to improve the fidelity and intelligibility of the received broadcasts in these bands which are prone to considerable fading, interference and noise. Transmissions in these bands typically have a low fidelity due to the restriction of a narrow bandwidth than broadcast FM transmissions on the very high frequency (VHF) band. A medium-wave broadcast station is restricted to a 9 kHz bandwidth whereas the bandwidth of broadcast FM station may be 75 kHz. Regarding the case of the medium-wave band, the challenge lay in creating a means of conveying high quality sounds to a wide audience using just a 9 kHz bandwidth. DRM allows the existing long-establish and reliable AM transmission infrastructure to be retained and re-used in a cost-effective manner for delivering much higher quality information to a mass audience.

The DRM system may choose one of a number of modes of transmission that best suit the propagation characteristics. One of the parameters in each mode is the type of guard-interval used between the COFDM symbols. These different modes use guard-intervals of different lengths between COFDM symbols depending on the propagation pattern. A long guard interval helps reduce ISI from the many different multi-path signals that may result from long-distance sky-wave propagation while a shorter guard-interval is suitable for local or regional coverage using ground-wave propagation.

A large number of evenly spaced sub-carriers are used to generate the COFDM symbol. The modulation schemes employed are Quadrature Amplitude Modulation 4-QAM, 16-QAM or 64-QAM. Guard intervals are appended to the transmitted symbols in order to ensure a high degree of resistance to inter-symbol interference due to multi-path fading effects caused by sky-wave propagation. The encoded information is conveyed using a similar process of sub-carrier insertion and multiplexing to form a sequence of COFDM symbols. Three different audio coding systems may be used to facilitate the different broadcasting requirements.

The DRM radio systems convey digital audio in the form of multi-lingual speech and music in addition to regular scheduled programming material. Images, news and other messages may also be transmitted to a very wide audience. Reception of these transmissions is more reliable than normal analogue techniques as frequency-diversity helps to combat frequency-selective fading and interference, inter-leaved data techniques help prevent loss of information from flat-fading effects and noise, and convolutional coding offers a level of error-correction to create a rugged and relatively error-resistant broadcasting system. The entire bandwidth of the channel may be used for data transmission or divided up into sub-bands, with each sub-band capable of carrying different sources of information or data types. In addition to providing near-FM quality audio, the DRM system has the capacity to integrate data and text, a concept not possible with traditional analogue transmission schemes present on the long, medium and shortwave bands. This additional content can be displayed on DRM receivers to enhance the listening experience and provide information regarding the programming schedule and even the location of the broadcasting site.

3.6.1 Services offered by DRM

This section examines some of the services that are possible using DRM, thanks to the service-multiplexing capabilities enabled by the use of OFDM.

3.6.1.1 Multiple Programme Delivery

As mentioned previously, using DRM it is possible to transmit more than one programme service at a time. The DRM system enables up to four different services to be transmitted simultaneously in the form of a multiplexed signal. The trade-off is that these services are limited by the quality and robustness requirements of the broadcaster in addition to the capacity of the multiplexed sub-carriers. For example it might be possible to transmit a high quality audio programme alongside another audio service delivering the latest news headlines. One other option is the simultaneous transmission of four speech services in four different languages instead of having to schedule each different language service for different segments of the daily broadcasting schedule.

3.6.1.2 Other Data-type Services

In place of an additional audio programme service, the broadcasting station may wish to transmit content which may be unrelated to the audio programme itself. An example of this type of information is text messages and/or advertisements. These messages may use a data rate of 80 bps, so therefore they do not consume a great deal of the available resources. These text messages are capable of adding extra value to the audio service in terms of newsflashes/new headlines and/or weather information. This information requires extra radio hardware such as a digital display and controller hardware in order to function as intended however. Alternatively this extra information may be used to control the user-device (e.g. automatic radio retuning or remote timing updates).

Automatic Frequency Switching

Automatic Frequency Switching (AFS) is a very useful technique that may be used in a DRM receiver to automatically choose the best quality version of the service being monitored by switching between the alternative frequencies that may be transmitting the same (or similar) content material. A list of the alternative frequencies in use is transmitted as part of the information contained in the DRM signal. Additional information such as programming schedules and regional information, such as when and where other sources of the same programme are available may also be transmitted. If applicable, this AFS list may also contain details of other services that may be received including analogue AM, FM and satellite-based DAB services. If the receiver is in the 'foot-print', or coverage area of two DRM signals of different frequencies, with both signals carrying the same content, then the DRM receiver may switch seamlessly between the two. It is also possible that the DRM receiver may switch between a DRM signal and an analogue AM signal if required. Another advantage of the DRM system is that broadcast stations may be selected by name/ID instead of frequency. This is a valuable feature as it enables the user to quickly browse the current programming being offered by the DRM stations within coverage.

Simulcast

Simultaneous broadcasting of two or more signals using a common frequency allocation is known as simulcast. In the context of DRM, this refers to the simultaneous transmission of both analogue AM and the DRM version using the same channel allocation. As a DRM signal uses OFDM, the power spectrum is relatively uniform across the channel bandwidth. A non-DRM receiver monitoring this signal may hear a noise-like signal. It is possible to

combine DRM and normal analogue AM transmissions using the same channel allocation. A receiver monitoring the analogue AM broadcast would notice an increase in the background noise level but the audio content should still be intelligible. A DRM receiver would be able to produce a higher quality audio using the DRM signal also present on the channel.

3.6.2 Design of the DRM system

This section examines some of the signal processing techniques used in the DRM system in addition to describing the coding and modulation processes. The first stage that will be described is the DRM transmitter, as shown in Figure 3.8.

The information source required to be transmitted using the DRM system may be audio, news, or control data, arriving from a broadcast studio. This information must be encoded and converted to DRM data frames according to the DRM specification. The structure of a data frame may also include training symbols used to align the receiver and compensate for channel conditions and extra information regarding the modulation scheme used and the sub-carrier indices that contain audio, images, or control information. This DRM frame is then converted to an OFDM waveform before being up-converted, amplified, and transmitted using the AM transmission hardware. The individual information services may be allocated to specific sub-carriers and multiplexed forming a sequence of OFDM symbols. The DRM system uses two classes of basic information called the Main Service Channel (MSC) and the Fast Access Channel (FAC).

3.6.2.1 Main Service Channel (MSC)

Referring to Figure 3.7 for the DRM frame structure shown terms of time and frequency, and Figure 3.8 for the block diagram describing the signal processes required, the first type, or class of information contains the encoded audio and data which are combined in the main service multiplexer to form the Main Service Channel (MSC). The MSC contains between one and four streams. Each stream is divided into logical frames, where each frame is of 400ms duration, as shown in Figure 3.7. Audio streams comprise compressed audio and may optionally carry text messages. Data streams may be composed of up to four sub-streams consisting of information data sequences, or packets. The arrangement and usage of the available sub-streams is dependent on the broadcaster's requirements.

3.6.2.2 Fast Access Channel (FAC)

The second class is information that bypasses the main service multiplexer and forms what is known as the Fast Access Channel (FAC) and Service Description Channel (SDC). These channels are used to convey the identification and parameter selection information used for a transmission. Upon correct reception of these channels at the receiver, the correct decoding and information extraction parameters may be ascertained. There is one SDC in every super-frame. Each super-frame of duration 1200ms comprises three transmission frames. The FAC is integrated in every transmission frame and is used to quickly obtain the information necessary for the receiver to be able to demodulate the DRM signal such as the modulation scheme and transmission mode. This channel is similar to the Fast Information Channel (FIC) used in the ETSI DAB system previously described.

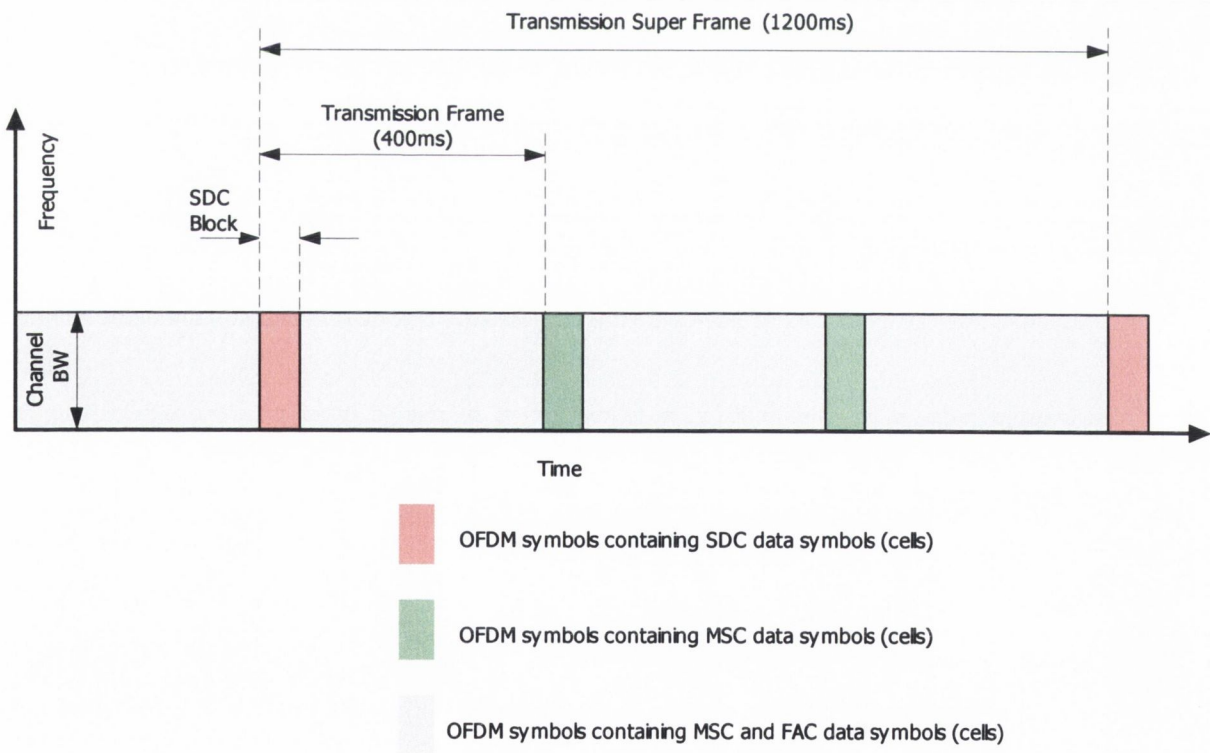


Figure 3.7: DRM frame structure in time and frequency.

Audio source encoders and the data pre-coders compress the raw input binary sequences into shorter signal sequences that may then be transmitted without requiring a very high data rate. The output of these encoders comprises two parts, each of which may be subject

to different protection levels during the subsequent channel encoder stages. The encoders used are:

- A subset of MPEG-4 Advanced Audio Coding (AAC) including error robustness tools for mono and stereo broadcasting.
- A subset of MPEG-4 Code Excited Linear Prediction (CELP) speech coder for error robust speech only broadcast in mono, for cases when only a very low bit-rate is available or especially high error robustness is required.
- Harmonic Vector Excitation Coding (HVXC) for the transmission of several speech only programs with a 4 kbit/sec bit-rate or a news channel accompanying the main program.
- Spectral Band Replication (SBR), which is an audio coding enhancement tool that allows the full audio bandwidth to be achieved at low bit rates. It can be applied to AAC, CELP and HVXC.
- Parametric Stereo (PS), which comprises an audio coding enhancement tool relevant to SBR that allows for stereo coding at low bit rates.

The multiplexer combines the protection levels of all data and audio services and inserts the information into the designated location in the frame structure of the DRM signal. Protection levels, or ratios, denote the type of error-protection coding used to ensure reliable delivery of the intended DRM signal. Transmitted AM signals may be subject to interference from DRM signals, DRM signals may be affected by adjacent, or co-channel AM signals, and DRM signals may be subject to interference from other DRM signals. Protection ratios are applied based on the anticipated source of interference. The DRM frame structure defines the arrangement of binary data from the encoders and extra information regarding the transmission modes and protection levels. An energy dispersal algorithm randomises the bits in a defined way in an attempt to reduce the possibility of spectral patterns, or unwanted regularity in the transmitted signal.

Extra redundant bits are added to the data in a defined way, using the channel encoder. This is done in order to increase the robustness of the data frame by providing a means for error correction. This channel encoder then maps the binary values to points on a QAM

constellation diagram. The bit-to-symbol-ratio, i.e. whether 4-QAM, 16-QAM, or 64-QAM is used, is defined by the mode of transmission required.

A pilot generator is used to enable the receiver to synchronise itself to the transmitter and also to estimate the effects of the channel in order to compensate using a channel equalisation stage. Known pilot symbols with fixed phases and amplitudes are used and the difference between the known pilot symbols and the received symbols is used to compensate for the effects of the channel. The different classes of data symbols, or cells, are gathered by the COFDM cell-mapping stage and allocated to sub-carriers. A DRM COFDM signal occupying a 10 kHz channel bandwidth may use between 88 and 226 sub-carriers, depending on the transmission mode.

The sequence of COFDM symbols is then created by performing an IFFT on the mapped waveform. The stage converts the digital representation of the COFDM signal into the analogue signal that will be transmitted. The transmission process simply involves replacing the traditional AM modulator audio stream with this DRM signal.

DRM offers four levels of signal robustness, which are used to help combat the time-variant effects of the channel. Noise, interference and fading are formidable factors in the long, medium, and shortwave bands. The transmission frame duration is the same for all four robustness modes, (i.e. 400ms) but the guard interval between the COFDM symbols has a different duration depending on the mode. Increasing the guard interval means that the multi-path signal components have a longer time to be attenuated and therefore minimising possible ISI effects. The trade-off is that the useful duration of the COFDM symbol is reduced as the guard interval increases. For local coverage using ground-wave propagation, a shorter guard interval may be used and thus an increase in the amount of data that may be transferred is possible. A table of the duration of the useful COFDM symbols, $T_U(ms)$, and guard intervals, $T_G(ms)$, for each of the four DRM robustness modes is shown in Table 3.2. The transmission frame duration, also shown in the table, is 400ms for each of the robustness modes.

Table 3.2: Table of useful COFDM symbol duration and guard interval duration for the four robustness modes used in DRM.

Parameters	Robustness Modes			
	A	B	C	D
$T_G(ms)$	2.666	5.333	5.333	7.333
$T_U(ms)$	24	21.333	14.666	9.333
$T_f(ms)$	400	400	400	400

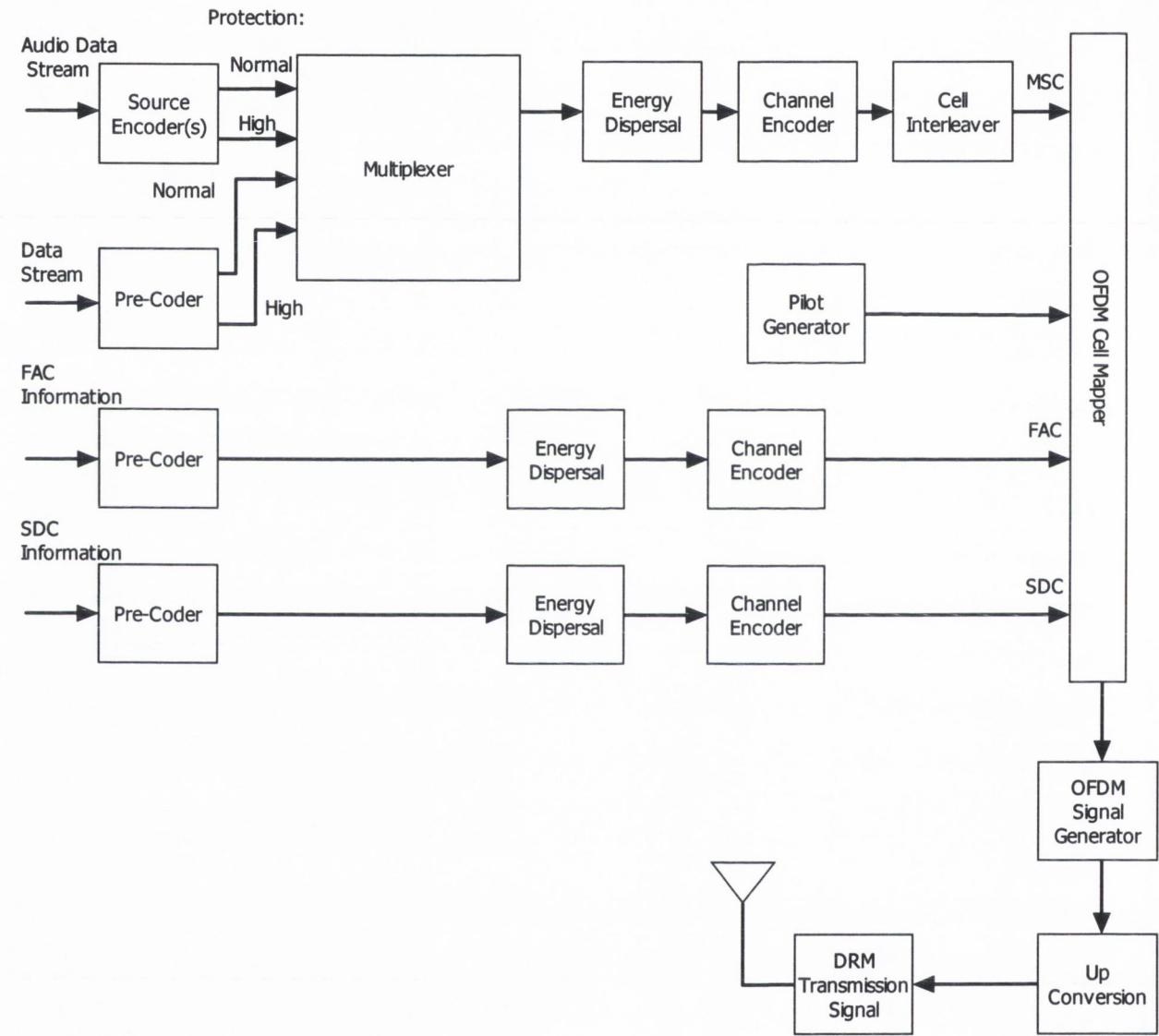


Figure 3.8: DRM transmitter block diagram and signal flow.

As mentioned previously, propagation in the long, medium, and shortwave bands can vary between ground-wave and sky-wave or combinations of both. Ground-wave propagation is adversely affected by electrical noise mainly, and signals propagating via sky-wave may suffer from additional factors such as multi-path delay, flat and frequency-selective fading, and Doppler fading effects. In some circumstances signals may reach locations in the

coverage area by means of both types of ground-wave and sky-wave propagation. In all of these cases the received signal robustness mode A is designed to deliver the highest bit rate possible within the context of local or regional service using ground-wave propagation. Robustness mode B is more suited for signals arriving at the receiver via sky-wave propagation. Where propagation conditions are more severe, such as for long paths with multiple hops, or near vertical incidence, where several very strong reflections may occur, robustness mode C or mode D may be required. In all these cases, either 64-QAM or 16-QAM for the Main Service Channel may be chosen, and this choice is mainly influenced by the expected Signal to Noise Ratio (SNR) expected at the receiver location. If the SNR is expected to be too low to support 64-QAM, the lower-complexity 16-QAM scheme may be chosen instead. The quality of the audio is affected by the chosen robustness mode and/or error protection required. The highest achievable bit-rate used for audio transmission is 24 kb/sec using a 9 or 10 kHz channel bandwidth. The level of error-protection required affects the maximum bit-rate, with a higher error protection and/or increased level of robustness resulting in a lower overall bit-rate.

3.6.3 DRM Performance

Three transmitters were made available for use by the DRM consortium members for field tests involving medium-wave and short-wave transmissions. Phase IIa of the field test schedule took place between July 17th and the August 14th 2000. Deutsche Welle operated a transmitter located in Sines, Portugal, and Merlin VT used two transmitters based in Orfordness and Rampisham, in the UK. In the case of the Orfordness transmission site, the average analogue AM transmission power was 250kW but the average digital AM (DRM) transmission power was 100kW. This large amplifier back-off was required to prevent the high Peak to Average Power Ratio (PAPR) of the COFDM signal from resulting in non-linear amplifier operation. The possible effects of such non-linearity would have been distortion and spurious emissions that may have resulted in interference to other users within the long, medium, and short-wave bands.

Five receivers were located in Germany, Netherlands, Cyprus and Finland using a variety of long wire antennas and dipoles. The tests involved initial measurements of the noise within the 10 kHz channel bandwidth in addition to measurements of interfering signals from adjacent frequencies. As of test of the quality difference between analogue AM and DRM, two four minute samples of audio and music were transmitted using DRM for test

sequence durations of 30mins. During this test, the transmissions switched between DRM and analogue AM for comparison purposes. The measurements involved subjective estimates of the signal strength (1=low, 5=high), interference (1=high, 5=none) and overall quality (SIO) (1=not intelligible, 5=excellent). The lack of DRM signal strength was the main cause for concern but a dramatic improvement over analogue AM reception was noted with improvements in interference and fading tolerability noted [DRMAnnex4D].

DRM is also of relevance to the work presented in this thesis as software techniques have been used to create and widely distribute a very useful and low-cost broadcast receiver. Software versions of a DRM receiver may be used to demodulate the IF signal of a DRM transmission and this DRM receiver application is available as download [DRMRX]. An open-source version of this receiver is also available and is created using C++ programming code [DReaM]. This receiver application in software approach is similar to the radio development and implementation that is focussed on in this thesis.

3.7 Relevant OFDM-based Products and Companies

It has been shown in the previous chapter that there has been a surge in the commercial availability of OFDM-based wireless communication products. This section describes some of the OFDM-based devices that are both currently available and are under development. These are mainly one-way information transfer systems targeted towards delivering high quality audio, video, images and news to a large audience. Emphasis is placed on producing affordable, portable, and efficient DAB receivers in addition to relatively easily distributable DRM receivers. WLAN transceivers based on the 802.11a and 802.11g standards described earlier are being used for high data-rate communications links to form networks of computers and PC-related devices.

The development trend being followed by many companies and manufacturers is to implement as many of the signal processing functions as possible using digital signal processors. A software version of a signal processing algorithm executed using a digital signal processor offers a reduction in the implementation and testing time required. As the signal processing chain is now in the software domain instead of relying on dedicated physical circuitry to perform the tasks, several advantages may be gained. Deployment, upgrades and management of software versions of the core processing functions normally

required in typical all-hardware transceivers is much faster, less complex, and less expensive than having to create new physical circuitry and hardware. The DRM receiver, for example is deployed using the internet. An approach such as this also maximises the re-usability of the supporting hardware such as the RF front-end, signal conversion hardware, and memory devices.

3.7.1 802.11 Products

The 802.11a standard previously explained is not backwards-compatible with 802.11b and does not offer the same communications range as its 2.4 GHz counterparts due to the propagation characteristics of the 5 GHz operating band that is used for 802.11a. The 802.11g standard, which is compatible with 802.11b but also offers a higher data rate for the same range as the latter, is attracting more attention with dual 802.11b/g devices being marketed such as the Proxim ORiNOCO [ProximOro] range of products. More recently, WLAN devices capable of implementing 802.11a/b and 802.11g (2.4GHz OFDM) in one package have been released.

Companies using the 802.11a standard in their product line, mainly offering WLAN cards for personal computers and laptops in addition to access points and Ethernet adaptors include Actiontec [Actiontec] which has a Personal Computer Memory Card International Association (PCMCIA) WLAN card using an Atheros AR5000 chipset for transceiver functions. Also on the list of 802.11a device suppliers are Intel, Proxim, and SMC. Intel [Intel2004] released the Intel PRO/Wireless series of 802.11a/b/g wireless network cards and access points. The highest data transfer rate of 54 Mbps was quoted to be achievable up to a range of 12 metres with the lowest data rate (1 Mbps) achievable up to a range of 90 metres. An extra feature of the Intel PRO WLAN card is the ability of the user to change the power emitted by the device in order to increase battery life but with a possible degradation in performance. As mentioned previously, Proxim has a range of 802.11 devices implementing each standard either singly or as combinations using one physical device. SMC [SMC2004] is another manufacturer offering wireless routers that use the 802.11g standard. A router passes data packets from one network to another with the objective of trying to choose the fastest route possible. The focus of the wireless market is moving towards wireless connectivity between laptops and fixed-Ethernet connections. OFDM in the form of the 802.11a and especially the 802.11g standards is proving to be a very popular choice thanks mainly to its high data rate capabilities and the 802.11b backwards-compatibility factor with 802.11g.

3.7.2 RadioScape

RadioScape [RadioScape] is a UK based wireless communications company that specialises in creating DAB broadcasting and receiver modules used for portable DAB receivers. These products are designed to receive the DAB signals described previously in Section 3.4. The products developed by this company are very good examples of the immediate advantages that may be gained by converting a lot of the typical hardware-centric radio functions to the software-only domain. RadioScape's main area of specialty is creating software solutions for DAB. The modules are based on a software-defined radio platform, which is essentially the normal RF front-end of a radio using a Digital Signal Processor and memory-storage to execute the majority of the signal processing tasks in software instead of a traditional all-hardware radio approach.

This DSP may be programmed by downloading the signal processing algorithms in programme form onto the device. One of the more interesting features with the receiver products created by RadioScape is the option of storing a segment of the broadcast in solid-state Flash memory, which is a type of Electrically Erasable Programmable Read Only Memory (EEPROM). This memory storage facility enables 'rewind' and 'pause' functions which have been facilitated by carrying out the signal processing functions in software. Broadcasting products are also available that enable the studio(s) to connect to the DAB ensemble provider using an Internet Protocol (IP) network connection. In fact, the software-defined radio platform that this is based on enables remote management and control of the broadcasting system.

3.7.3 Flarion Technologies

Flarion Technologies [Flarion] is another wireless communications company of particular interest to the work presented in this thesis. Flarion is based in New Jersey, USA and develops and produces mobile broadband WLAN devices using OFDM as the transmission technology as a bridge between fixed IP networks and mobile devices. The main features of their OFDM-based devices are that they are IP packet-based and use a variation of the standard OFDM presented in this chapter called Fast low-Latency Access with Seamless Handover OFDM (FLASH-OFDM™). This is essentially standard OFDM but regarding the wireless capabilities of the FLASH scheme, the difference is that the specific sub-

carriers used for transmission may change for each of the transmitted OFDM symbols. The aim of this frequency-hopping OFDM is to further reduce the effects of interference and multi-path fading. This may be achieved as the transmitted information is present on each sub-carrier for a reduced length of time than for normal OFDM implementations such as 802.11a/g. This process is similar to spread-spectrum and may be further described as fast-hopping OFDM or Dynamic OFDM (DOFDM), which is described in a later chapter in this thesis.

3.8 Conclusions

This chapter has been shown that OFDM is a very popular transmission technology and has several highly useful commercial applications. OFDM is used for high data-rate wireless communications links between two or more WLAN devices. High quality multi-programme audio services may be multiplexed as a sequence of OFDM symbols and delivered to a wide audience using DAB. The robustness and high data-rate capabilities of OFDM in a challenging wireless environment have resulted in the re-vitalisation of the long, medium and shortwave bands. Reliable and error-resilient near-FM quality digital audio, images and news may now be conveyed to a very large audience using these bands, even though the received signals may have been affected by fading, noise and interference. OFDM transmission technology is capable of being improved further by moving away from the traditional fixed-architecture concept of radio and progressing towards more organic flexible-architecture radio system design.

4

NEXT GENERATION OFDM IMPLEMENTATIONS

4.1 Introduction

In the last chapter, a variety of current implementations of OFDM communication systems have been discussed. OFDM is expected to evolve further as a dominant information-transmission technique for future wireless communication products. The potential for OFDM in the future is the focus of this chapter. The chapter begins by discussing developments that are on the horizon for OFDM-based communication systems. It is envisaged that reconfigurable radio will play an important part in the communication systems of the future. Reconfigurable radio is the topic of the second part of this thesis. Section 4.3 highlights the key technologies that have formed the evolutionary path of reconfigurable radio and identifies some of the main ways in which an OFDM transceiver may be significantly improved. Section 4.4 draws conclusions from this chapter.

4.2 Envisaged Long-term Prospects of OFDM

It is expected that OFDM will prevail as a high data-rate and robust transmission technology used for more forward-looking wireless communication products and ideas in the coming years. For example, the previous chapter described a proposed IEEE 802.11n standard that may be capable of allowing data rates of over 500 Mbps to be achieved. This is the expected performance of an OFDM-based technology in the short-term, and it shows that OFDM technology offers dramatically increased data transmission rates over single-carrier wireless communications systems. Of direct relevance to the work presented in this thesis is how the performance and capabilities of OFDM-related wireless communications technology may be improved even further in the long-term.

Apart from the RF hardware considerations, OFDM is highly suited for implementation in the digital domain and this is the key to how OFDM may evolve further. The OFDM-related products described in the previous chapter all share a view that the signal chain used in a radio is fixed and may not be modified easily once implemented to form a wireless device. The characteristics of a time-varying wireless communications channel and indeed the actions and behaviour patterns of the radio user are not fixed however. It is a logical step therefore, to envisage the creation of a transceiver that reflects the changes in the wireless communications channel environment and radio-user behaviour in the form of modifications to the radio. One of the main objectives of these modifications is to improve the performance of the OFDM transceiver by enabling the radio to adapt to the wireless channel environment and/or the requirements of the radio user. Focusing in the on the signal chain associated with an OFDM transceiver, this wireless channel adaptation procedure involves being able to modify the operation, characteristics and objectives of one or more of the individual signal-processing stages that comprise the OFDM transceiver signal-chain.

In summary, it is the view of the author that there are two main areas where OFDM may be improved. These areas relate to:

- Wireless communication channel considerations.
- User's transmission requirements.

4.2.1 Wireless Communications Channel Considerations

Regarding the wireless communication channel, each of the sub-carrier channels used for transmission may be considered as an independent time-varying channel. Therefore the characteristics of the channel environment associated with each of the sub-carriers may not necessarily be exactly the same. The amount of information that may be conveyed to a receiver through each of these channels is limited [Shannon1949][Nyquist1928]. As the wireless channel environment associated with each sub-carrier may not be the same, this means that the channel capacity associated with each sub-carrier may also vary with time. In terms of the wireless communications channel, OFDM transmission rates may therefore be improved by taking account of the possibly continually changing channel capacity of each of the sub-carriers. One method of improving the data-rate capabilities of OFDM may be to increase the transmission rate on those sub-carriers with a higher channel

capacity than the other sub-carriers. The transmission rate may be increased by employing a modulation scheme with a higher bit-to-symbol ratio. This means that more binary information may be represented using one modulated data symbol thus increasing the data rate.

In addition to the transmission data that contains information from the source, (e.g. voice, images, video, music, news), other data used by the destination receiver for synchronisation and control purposes are also transmitted. These transmissions, which may be collectively referred to as ‘transmission overheads’, do not contribute to the amount of data from the source received by the destination receiver. For example, consider the case where the sub-carriers in an OFDM symbol are modulated using different modulation schemes in order to increase the overall transmitted data rate. In order to correctly demodulate the received sub-carriers, the receiver must know how each of the sub-carriers is modulated. One method of achieving this is to transmit the modulation scheme information along with the OFDM symbol. This extra information does not contribute to the data received by the destination receiver and occupies spectrum that could be used to increase the amount of information transmitted from the source. An alternative method of increasing the amount of data transmitted therefore, is to reduce, or even eliminate receiver-synchronisation transmissions.

This approach places the onus of synchronisation and interpretation of the incoming data onto the receiver. In this scenario, the receiver must be capable of performing ‘blind synchronisation’. Blind synchronisation is the term that may be used to describe a receiver that achieves synchronisation and correct interpretation of the received signals without *a priori* knowledge of the transmission parameters and configuration.

In terms of OFDM communications, one method of blind synchronisation may be to determine what modulation scheme is being used for each sub-carrier in a received OFDM symbol. This may be achieved by analysing the characteristics of the signal waveform and estimating the modulation scheme from that information. A second method of blind synchronisation may be to determine where the start of an OFDM frame occurs in an intercepted OFDM signal sequence. As discussed in Chapter 2, the receiver requires knowledge of the start of the OFDM frame in order to correctly extract the OFDM symbols contained in the frame.

Consider the case where the wireless communications channel is crowded or affected by strong interference. If a traditional transceiver attempted to transmit information using the affected carrier frequencies, there may be a high possibility that this information would not be received correctly by the destination radio. The robustness of OFDM may be therefore be increased by enabling the OFDM transmitter to avoid using the affected sub-carriers. As details of the affected frequencies may not necessarily be predicted, this concept relies on the ability to inform the destination receiver of which sub-carriers are being used. A means of achieving this without having to increase synchronisation-related transmissions is desirable.

4.2.2 User's Transmission Requirements

Traditional radio design has afforded little flexibility to the user in terms of choosing a data transmission scheme that best suits their requirements. The user may want to convey information to a destination radio using the highest data-rate possible or using the most cost-effective mode of wireless transport. In general, traditional radio design limits the user to a choice between pre-selected transmission modes, frequency bands and transmission power. Future radio possibilities may include the ability of the radio to automatically choose a transmission mode or frequency band that best suits the user's objectives. The feasibility of these ideas depends on the ability of the radio to modify the signal chain of the radio, or indeed, replace the entire signal chain with another radio implementation, if required. Therefore, the long-term prospect of creating a radio that can automatically adapt to the user's wireless communication needs depends on the ability of the radio to modify the core signal-processing architecture of the radio. In other words, adaptation of the radio to the user's requirements and spectrum-usage preferences may not be possible unless the radio is capable of automatic and dynamic adaptation to the wireless channel.

4.2.3 Fundamental Radio Design Alterations

The ability of a radio to adapt its operational characteristics and channel access profile to the time-varying characteristics of the wireless communications environment is largely determined by the radio architecture. As stated previously, the radio signal chain must be capable of being modified as the characteristics of the wireless channel vary. Fixed radio architecture, synonymous with traditional hardware radio design, is not conducive to the

realisation of a transceiver capable of automatic re-organisation or incremental modification of the core signal-processing stages comprising the radio signal-chain.

The solution therefore, is to create a type of radio where the signal chain, radio structure and even the radio application may be automatically reconfigured. This ‘reconfigurable radio’ solution may then enable the realisation of wireless channel adaptation. Looking further ahead, this reconfigurable radio may then act as an enabling technology for the creation of a radio that automatically caters to the objectives of the user.

4.3 Reconfigurable Radio – An Enabling Technology

The goal of creating an improved and more useful radio may be reached by taking an ‘organic’ approach to radio design where the internal structure of the radio is capable of being dynamically modified, augmented, reduced, re-ordered and improved. A radio with these capabilities may be referred to a reconfigurable radio. This reconfigurable radio approach is therefore expected to be an integral part in the evolution of advanced communication techniques in the future. In the context of this thesis, reconfigurable radio is expected to play an important role in advanced OFDM communication system designs. In order to fully understand what is meant by reconfigurable radio before progressing to explanations of how it enables OFDM to be improved, it is useful to describe the stages of radio development leading to highly reconfigurable radio designs.

This section therefore describes how radio design principles have progressed from all-hardware fixed radio architectures to hybrid software/hardware radio designs. One of the motivational points of this approach is that reconfigurability options increase further as more of the radio is implemented using software. In order of increasing reconfigurability options therefore, the description of programmable radio leads onto software-defined radio. By moving as many of the hardware radio entities as possible into the software domain, a software radio may be created. Software radio can be used as a platform for the creation of a reconfigurable radio. Cognitive radio, which is an advanced type of reconfigurable radio, is also discussed. For more in-depth understanding of the related background material, the reader is directed to Pereria’s [Pereira1999], Kohno’s [Kohno1999] and Mackenzie’s [Mackenzie2004], discussions of software-defined radio, software radio and related development trends.

4.3.1 Programmable Radio

Historically, radios have been designed with very little functionality other than the basic power on/off, audio output level control, a receiver threshold control (squellch), and a method of switching between the available RF channels or frequency bands capable of being accessed by the radio. The entire radio structure of these devices is hardware-based, the functionality of the radio is defined by physical circuitry, and the performance of the radio is largely dictated by the characteristics of this physical circuitry. If a new radio function or feature is desired, then extra hardware, which implements the desired function or feature, has to be incorporated into the radio. This may be achieved by either attaching an extra hardware module or even completely replacing the existing radio hardware with a new version that includes the new features. One major drawback with this is that a radio with a greater physical size may ensue and the process may be costly in terms of development, manufacturing, installation and testing new hardware. A modular radio structure, where extra bands/modes of operation may be implemented using external modules, offers some measure of adaptability but as discussed in the previous chapter, this method is not ideal. The reasons for this are that the radio user has to carry this extra hardware, physically modify the radio themselves and these hardware modules may be prone to malfunctions. For most commercial transceivers however, new features mean that a new generation of the radio must be manufactured. The implications of device obsolescence are that there are few, if any, possibilities for re-using the obsolete hardware and it is often common that a brand new transceiver generation may not be backwards-compatible with the previous series of radios that may still be in service. Advances in signal processing techniques and digital signal processors have resulted in the ability to implement sections of the radio as algorithms stored in memory and running on a Digital Signal Processor (DSP) instead of the traditional hardware approach of increasing the complexity of the radio circuitry. The area of software-controlled hardware or programmable radio has formed the transition region between fully-hardware and software-dominant transceiver systems. In the late 1970's, initial programmable radio technology was largely driven by the United States military's objective of creating a single radio capable of interoperability with all of the military radio models, modes of operation and frequency allocations [Camana1988]. Programmable radio is a radio system implemented in hardware, but instead of traditional circuit components (e.g. resistors, capacitors, inductors) dictating the function of the radio, the main functions of the radio are implemented using Integrated Circuit (IC) technology. Extra radio modes of operation or

access to other frequency bands required additional IC-based radio modules or new ICs with the additional radio features incorporated into the original design.

4.3.2 Software Defined-Radio

The next stage in the software radio evolutionary ladder after programmable radio hardware, is Software-Defined Radio (SDR). A software-defined radio is a radio system that may be programmed to implement many different radio standards and user-defined applications. The signal-processing chain in a software-defined radio comprises a mix of programmable hardware using Field Programmable Gate Arrays (FPGAs) and software algorithms operating using a Digital Signal Processor (DSP).

A significant amount of SDR-related research and development regarding has occurred since the first major U.S government and U.S military SDR project called SPEAKeasy commenced in the early 1990's [Torre1992] [Lackey1995] [Vidano1997]. The original main objective of this project was to enable a number of military radios to be implemented on a single SDR device in an attempt to reduce military spending and increase interoperability options between the numerous different military radio services. The Joint Tactical Radio System (JTRS) Joint Program Office (JPO) also commenced development work on a SDR-based Software Communications Architecture (SCA), which originally had similar objectives as the SPEAKeasy project [Place2000] [Melby2002]. The majority of the SDR implementations constitute combinations of DSPs and one or more FPGAs that control the RF front-end. The operation of the SDR defined is defined in software using a high-level programming language or Assembly code, which may then be compiled, built and executed using a DSP. Radio hardware configuration is performed using software, which stores essential operational information in reserved memory locations. An example of this is a SDR implementing decimation and filtering using a FPGA where the decimation factor and filter taps are defined by storing the relevant values in a series of memory locations specific to this key radio components. This information effectively defines the operating characteristics of the hardware used in a SDR while the rest of the radio objectives are realised using software algorithms operating on a DSP. In the case of a typical SDR therefore, the SDR hardware may perform the channel selection, amplification, filtering and decimation/interpolation before producing either a complex-valued, magnitude and phase, or frequency and amplitude signal sequence. The DSP then enables the remainder of the signal-processing tasks to be carried out using a software algorithm.

4.3.3 Software Radio

Moving on from software-defined radio is a type of radio that apart from the antenna, initial signal amplification and signal conversion to/from analogue and digital domains, is implemented in software. Joseph Mitola [Mitola1992][Milola1995], a consulting scientist involved with the U.S. Department of Defence claims to have defined the term 'software radio' in 1991 when he was a chief scientist working with Electronic Systems E-Systems at that time. His definition of software radio was:

A software radio is a set of Digital Signal processing (DSP) primitives, a meta-level system for combining the primitive into communications systems functions (transmitter, channel model, receiver...) and a set of target processors on with the software radio is for real-time communications [Mitola92].

It is important to point out that the boundaries between SDR and software radio are not clearly defined in the aforementioned publications by Mitola and Lackey in addition to Reed [Reed2002] and further discussions regarding the contrasting views may be found by consulting Mackenzie's doctoral thesis [Mackenzie2004].

Unlike software-defined radio, as much as possible of the channel-selection, decimation/interpolation and filtering operations may be carried out in software using a software radio. An radio implementation approach leading to this method was first taken by the Collins Defence Communications Division (CDC) of Rockwell International in 1990, to create a modular radio, with a configuration defined by software [Roggow1990] [Church1990]. This was named the Spectrum 2000™ Radio System. It was only a short conceptual step between placing radio modules in a computer and actually implementing the functions of each hardware radio module in software using the computer processor, to effectively create a software radio.

An 'ideal' software radio is a radio system that operates entirely in the software domain. Some essential hardware is required however. This includes the antenna, RF to IF (or RF to baseband) conversion, filtering and amplification stages, and ADC/DAC stages, in

addition to the digital signal processor itself. An ideal software receiver does not require an RF to IF (or RF to baseband) conversion stage but instead it digitises and demodulates the signal directly from the antenna. An ideal software transmitter converts a digitally generated signal directly to an analogue modulated RF signal, which may then be transmitted by connecting this signal to the antenna via RF front-end amplification and up-conversion hardware. As the modulation and demodulation radio functions are performed in the software domain, this allows the implementation of efficient and creative modulation, demodulation, and radio spectrum access schemes using a digital signal processor without having to add any extra hardware to the software radio system.

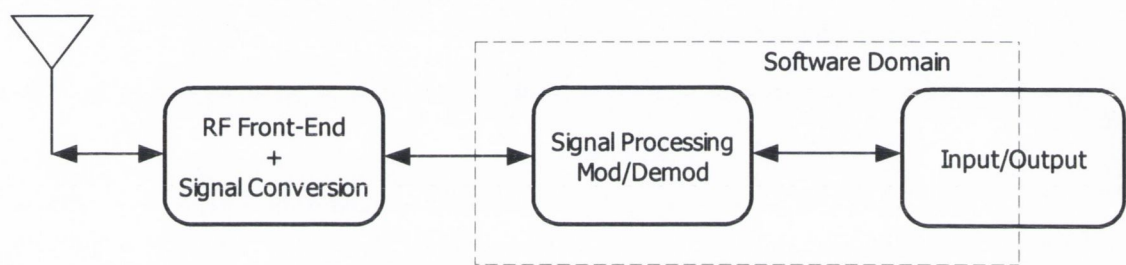


Figure 4.1: Block diagram of a transceiver showing the basic processing blocks

As previously mentioned, a practical software radio transmitter relies on hardware devices such as the RF front-end, known as the air-interface, to convert a digital signal into an analogue electromagnetic signal, which may then be amplified and transmitted. A conceptual diagram of this is shown in Figure 4.1 where the main stages of a transceiver are illustrated. The region where the radio may be implemented purely in software is enclosed using a dashed line. A transmitter requires an RF front-end to amplify, filter, convert the signal from digital to analogue domain and perform up-conversion to the required band. Similarly for the receive process, an intercepted analogue waveform from the antenna must be amplified by the air-interface, down-converted to IF or baseband and then converted to a digital waveform. This digital signal sequence may then be demodulated by carrying out the remainder of the signal processing operations in software. This stage is shown as the middle stage of Figure 4.1. An input/output stage acts as the connection between the radio and the information sources/sinks. This is illustrated using a dashed-line in the figure as being a half software, half hardware process as physical input/out devices such as data displays, audio output devices and Ethernet connections connected to the radio using software.

In order to summarise this discussion so far, it is useful to state what is understood by the term 'software radio' in the context of this thesis:

A practical software radio is where the modulation, demodulation and waveform generation processes are defined and exist as software entities, and operate on digitised signals using a digital signal processor. Analogue to Digital and Digital to Analogue Converters (ADC/DAC) form the boundary between the digital and analogue signal domains. Signal amplification and frequency-translation to/from the band(s) of interest are carried out using a wideband hardware RF front-end connected to one or more antenna(s).

This view of software radio has also been taken by the SpectrumWare group [Tennenhouse1995] [Tennenhouse1996a][Tennenhouse1996b] and Vanu Bose [Bose1999a][Bose1999b] who founded a company named Vanu [VanuInc] based on his work within the SpectrumWare group.

A software radio may have multiple RF operational characteristics depending on the modulation/demodulation schemes and operating parameters specified by the designer. Specifically, this means that a software radio may be comprised of a collection of functional tasks, RF interface access methods, and multiple channel coding techniques available for either a single software radio application, or as part of multiple simultaneous software radio applications. Simultaneous applications are two or more signal processing algorithms may operate at the same time. An example of this is a software radio that may be required to operate in two different frequency bands using different wireless communication techniques. An example of this scenario therefore, is a Binary Phase Shift Keying (BPSK) receiver operating on the 900 MHz Industrial, Scientific and Medical (ISM) band in conjunction with an OFDM transceiver operating on the 2.4 GHz ISM band. One wideband RF-interface, capable of transmitting and receiving signals using these two frequency bands, in addition to wideband ADC and DAC are used and shared between the independent applications. This example is only feasible by using a digital signal processor with multi-tasking (or multi-threading) capabilities.

Many of the aspects of the RF interface can be defined in software in a software radio, which enables an ideal RF-interface to be independent of the actual radio implementation. Ideally, a universal extremely wide-band RF front-end and antenna system means that any frequency of interest within the bounds of the system may be accessed in both receive and transmit modes of operation. The physical-layer abstraction methods used in the 802.11 WLAN standards, which were described in the previous chapter, are good examples of early attempts to achieve this goal. Waveforms can be generated to suit that user and/or channel characteristics, the channel access schemes and the configuration of the multiple radio personalities that can access the air interface simultaneously. A software radio can be reconfigured using a dynamic parameter based Man-Machine Interface (MMI), or automatically by the radio itself to facilitate the creation of a reconfigurable radio.

4.3.4 Reconfigurable Radio

Reconfigurable radio is an extension of software radio. A reconfigurable radio is a software radio where the transceiver characteristics and objectives may be dynamically modified by the user and/or by the radio itself by reacting to information from one or more external sources and/or the receiver. Reconfigurability is a means of improving transceiver design by enabling the radio to adapt to the time-varying wireless communications channel conditions and user-requirements. Pereira defines reconfigurability as:

“Re-configurability comprises...the pervasive use of software-re-definition (it is not enough to define once), empowering (possibly live) upgrades/patching of any element in the network, and of all services and applications running on it.” [Pereira1999]

Reconfigurability is rapidly becoming an integral part of commercial wireless communication devices. Bucknell [Bucknell2000] presents experimental results regarding ‘on-the-fly’ user-reconfiguration of a DECT platform and proposes an Over The Air (OTA) reconfigurability by download scheme. O’Mahony and Doyle [O’Mahony2002] share the view that 4th Generation (4G) wireless networks will feature software radio as a core component of the physical layer. Pereira [Pereira2000][Pereira2001] envisages that reconfigurable radio will play a key role in an envisioned (4G) wireless network. Palicot [Palicot2003] proposes a reconfigurable receiver called a Self Adaptive Universal Receiver (SAUR) capable of demodulating several wireless communication standards including

DAB, GSM and DECT. In 2003 also, Helmschmidt, Shüler *et al* [Helmschmidt2003] investigated how a wireless device may be reconfigured to implement OFDM and Universal Mobile Telecommunications System (UMTS)/Wideband-Code Division Multiple Access (W-CDMA) reconfigurable receivers for 3rd Generation mobile communications, however the proposed radio platform uses a combination of non-dynamically-reconfigurable DSP, FPGAs and Arithmetic Logic Units (ALUs) thus may be classed as a SDR.

The communications industry is motivated towards creating radios that can handle multiple frequency bands, interpret the many transmission protocols that are used on all of these frequency bands, be reconfigured on demand, and allow simple upgradeability options. Reconfigurable radio technology has been driven by the need to control and coordinate the many different capabilities possible with present commercial wireless end-user terminals. Application and operational software account for most of the software present in these devices and reconfiguration is playing an increasingly important part in making remote device upgrades using downloaded software radio components possible. Regarding more customer-focused radio issues such as Quality of Service (QoS) provisioning in reconfigurable radios, Aghvami, Le *et al* [Aghvami2001] present an analysis of how this may be achieved using a reconfigurable radio. Looking further ahead, Faroughi-Esfahani and Vardoulas [Faroughi-E2001] propose a power-control-based means of locating and removing rogue software-reconfigurable wireless devices affected by un-recoverable radio malfunctions or even corrupted by software virii from a network. Hoffmeyer, Park *et al* [HoffMeyer2004] investigates the mechanisms that are required for OTA software download for reconfiguring wireless devices, and Paillassa [Paillassa2003] described a proposal for satellite reconfigurable radio. Although the focus of these last four papers is on SDR technology, it shows that radio reconfiguration and the benefits a user may gain from such a technology have attracted commercial attention. Reconfigurability is not just a means to help improve usability and communication capabilities, however. OTA downloads and self-reconfiguration may also be used to automatically correct defects in, and upgrade the signal-processing software core of a reconfigurable transceiver.

The vision of reconfigurable wireless communication devices presented in this thesis concerns the application, services, and control layers also. However, to a much a greater degree, the focus is on lower-level physical layer reconfigurability in terms of leveraging the information regarding the wireless communication channel environment and usage

patterns to optimise, adapt and attempt to obtain maximum power efficiency, maximise channel capacity usage and maximise the signal processing efficiency using a reconfigurable radio. Shen, Su *et al* [Shen2003a] [Shen2003b] state that four different layers of reconfigurability exist with a reconfigurable radio using a GPP. They are classed as soft-control, parameter, function and physical layers. In this thesis, using the IRIS reconfigurable radio platform, a similar distribution of reconfigurability tasks exist. These are classed as application reconfigurability, radio structure reconfigurability and radio component reconfigurability.

4.3.4.1 Application Reconfigurability

In this thesis, the term ‘application reconfigurability’ refers to reconfigurable radio applications that may be reconfigured to form a different radio implementation by rearranging the radio structure to adding more, or removing, signal-processing stages from the radio structure. An application may be reconfigured by either swapping the entire application software for the software implementation of the new application, or interchanging processing modules or signal-processing stages within the existing software resulting in the new radio implementation. One example of this is the 900 MHz ISM-band BPSK receiver scenario referred to earlier; a reconfigurable radio implementation of this application would enable dynamic reconfiguration of the radio application in order to create an implementation of a 2.4 GHz ISM-band OFDM transceiver. This example of radio reconfiguration manifests itself as a software modification if a RF front-end with transmission and reception operational capability is employed. In other words, two or more different radio applications may be created without having to physically modify the hardware.

A second example involves changing the objective of an application that may be embedded in a much larger radio application. For example, consider a reconfigurable radio application comprising two software modules (or stages) as shown in Figure 4.2. The first functional module is a low-pass anti-aliasing filter and the second module is a decimation stage. The first application in this example is a down-sampling, or decimation application. The specific task of this application may be reconfigured to realise an up-sampling (interpolation) operation by firstly, replacing the decimation module with an interpolation module, and then placing the low-pass filter after the interpolator. This example involves

removing (or deactivating) one signal-processing stage, inserting a different signal processing function into the signal path (in this case, an interpolation stage), and rearranging the order of the filter stage (place it after the interpolation stage). This second application is also shown in Figure 4.2. Again, this is all accomplished in software and may either be initiated by the user, or automatically by the radio control system.

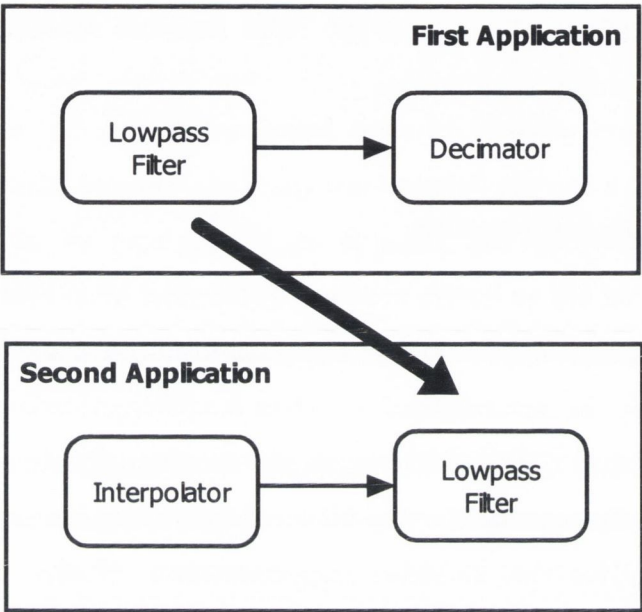


Figure 4.2: Concept of application reconfiguration by replacing and re-ordering software-processing modules.

4.3.4.2 Radio Structure Reconfigurability

A reconfigurable radio structure is closely related to the reconfigurable applications concept described in the previous section. Reconfiguring the radio structure may not necessarily result in a complete change of application however. A reconfigurable radio application comprised of sequential signal-processing stages, each accepting one or more inputs, executing a specific set of tasks (or algorithms), and then passing one or more outputs to the following stage, may be restructured on the fly if required. This means that both the order of the existing signal-processing stages and even each individual stage may be changed to improve the performance of the radio. This is extremely useful for changing internal processes, such as replacing a filter for a more efficient or effective design. The motivation for this is to improve the performance of the radio or reduce unnecessary power consumption due to a processor-intensive task when it is not required. The objectives of the application are not changed; only one or more signal-processing stages within the application processing chain are modified.

4.3.4.3 Radio Component Reconfigurability

The highest level of reconfigurability exists where the parameters of each signal processing component may be modified. This lower level of reconfiguration allows the user or system to change features of the components based on input regarding the channel quality, Bit Error Rate (BER), spectrum usage or co-channel/adjacent channel interference, or the user's requirements. For example, consider the modulation stage of a reconfigurable radio transmitter; a means of reconfiguring this stage would enable the modulation scheme to be changed dynamically in order to increase or decrease the transmitted data rate. The estimated Signal to Noise Ratio (SNR) varies with time therefore when the SNR has increased beyond a specific threshold, the channel may be capable of supporting a more complex modulation scheme thus increasing the transmitted data rate. The reason for reconfiguring the modulation stage in this way would therefore be to attempt to achieve maximum channel capacity usage in a time-varying wireless channel environment. This is one simple example of how being able to reconfigure the individual signal-processing stages (or referred to as Radio Components) in a reconfigurable transceiver.

4.3.4.4 Reconfigurable Processors

In 2002, Gray [Gray2002] described a reconfigurable processor prototype implementing physical, data link, and transport layers, which was demonstrated in the Jet Propulsion Lab, California. The reported physical layer is a BPSK radio. The actual demonstrated reconfigurable system aspects are not clearly defined, however. The original aims of this work were to allow the modification of the coding and modulation scheme used, for weather-affected communication links and roaming receivers. Working on the assumption that next-generation network clients, applications, or service, will choose network resources based on their data transfer demands and/or the current channel capacity, Nokia's German Research Centre proposed approaches described by Brakensiek, Oelkrug *et al* [Brakensiek2002] enabling this feature using a software radio platform. Brakensiek's paper outlines the reconfiguration options and approaches for dealing with a heterogeneous communications network. After initial configuration in the post-manufacturing stage, three other major classes of possible reconfiguration were defined by the authors

- Reconfiguration with downtime.

This is expected to be an infrequent occurrence, will take a considerable more lengthy time to complete with the unit powered down.

- Reconfiguration on a per call basis.

This is dynamic reconfiguration process, incurring a brief downtime, and activated on a per-call basis.

- Reconfiguration per timeslot.

A highly frequent level of reconfiguration involving several changes to the parameters and structure of the radio.

The approach and visions presented in Brakensiek's paper are limited however. Emphasis is placed on implementing the reconfigurable radio as a DSP and FPGA hybrid. Only one terminal or wireless standard may be utilised at a time. This thesis is based on a vision that future network users will be able to send and receive data of any sort to multiple recipients, and using several different standards simultaneously. A multi-threaded reconfigurable radio platform allows many radio implementations to operate simultaneously using a single RF front-end, effectively creating a shared RF spectrum-access communications terminal.

Classification of the different levels of reconfigurability plays a key role in the creation of a reconfiguration management structure. High-level, but infrequent and lengthy reconfiguration is expected to result in downtime for the radio device, or terminal. The expected downtime, (i.e. the length of time that the device would be out of service) is inversely proportional to the degree of radio modification, or reconfiguration granularity size required. The obvious advantages of using a GPP platform for reconfigurable radio have lead to proposals for management structures for the co-existence of multiple radio standards using a single GPP-based terminal [Shen2003a]. The system summarized in this paper is expected to be able to offer multimedia and internet support using a Layered Reconfiguration Management controller that creates new radio structures and memory allocations, determines what hardware resources are available, in addition to the destruction of redundant processes and de-allocation of memory. Based on the research work presented in these examples, the ultimate reconfigurable radio is expected to have an

organic and evolving lifecycle in addition to being independent of the signal-processing platform it is implemented on.

4.3.4.5 Reconfigurable Radio in this Thesis

Based on the discussions of reconfiguration and reconfigurable radio in this chapter, it is envisaged that reconfigurable radio will enable a new and advanced generation of wireless communication devices. Before progressing any further in this thesis, it is important to define what a reconfigurable radio is understood to mean in the context of this thesis. The term ‘reconfigurable radio’ is developed from the author’s definition of a software radio defined in Section 4.33, and is therefore understood in this thesis to mean:

A practical reconfigurable radio is a software radio where the radio application objectives, signal-processing stages comprising the signal chain and parameters associated with these signal-processing stages may be dynamically modified, replaced, or augmented. The stimuli for this radio reconfiguration are the characteristics of the wireless channel environment and/or user(s) wireless access requirements.

4.3.5 Cognitive Radio

Cognitive radio is an extension of reconfigurable radio and represents one direction in which the reconfigurable radio techniques presented in this thesis may evolve. Cognitive radio comprises a wireless transceiver that not only is capable of extracting information from the receiver regarding the current wireless channel environment, but is capable of decision-based autonomous control and reconfiguration using information regarding its current location, device owner preferences and usage-history patterns, the time of day, external temperature and amount of daylight, available power in addition to adhering to a radio and user operational guidelines. The term ‘cognitive radio’ was coined by Mitola [Mitola1999] in 1999 to describe a reconfigurable radio capable of choosing a course of action based on observations of not only the wireless channel environment, but also based on the device user's present, past and anticipated actions. Mitola defines cognitive radio as follows:

"Cognitive Radio refers to a class of software radio that employs model-based reasoning and at least a chess-program level of sophistication in using, planning, and creating radio etiquettes. Cognitive Radio is a goal-driven framework in which the radio autonomously observes the radio environment, infers context, assesses alternatives, generates plans, supervises multi-media services, and learns from its mistakes" [Mitola1999].

A cognitive radio may include several sensors which continuously monitor many aspects of the local environment. Examples of such conditions include the location of the device, spectrum-usage restrictions in the current location, current power reserves, anticipated user actions and other environmental factors including the time of day, current light level and local temperature. The concept of this cognitive radio system also includes the ability to conform to spectrum etiquette. This means that the radio could conceivably vary its transmitted power output level if it detects that interference is being caused to other legitimate users, or if local spectrum regulations that may be in service dictate a maximum output power level. One of the future possible advantages of cognitive radio is expected to be a 'spectrum-pooling' capability, where the cognitive radio could acquire a section of the frequency spectrum according to the priority of the request. The realisation of this vision would enable different emergency services, converging on the scene of a disaster to automatically converge on a common shared spectrum allocation also in order to allow seamless interconnectivity between all the different response groups. This would be achieved by the collaborative efforts of all of the cognitive radios involved.

Among Mitola's ideas for a cognitive radio is the development of a greater attachment or 'bond' to the device-owner. This means that the device could 'learn' the personal radio habits of a user and tailor itself to best suit the person. Another simple proposal by Mitola is that the radio could detect that it is in the user's hand by comparing the difference between the temperature of the back panel and the front panel of the radio. If the device is being held in the hand, the temperature of the back panel would be significantly greater than that of the front panel. A sudden shock coinciding with a rapidly decreasing temperature measured on both panels could indicate that the device was dropped. In this case, a cognitive radio would attempt to alert the user by sounding an alert, or by

conveying a message to a colleague, or even initiate a course of action that would result in the safe recovery of the device.

A realisable cognitive radio would require significantly more signal-processing power than is currently available. The radio platform would have to be capable of extreme reconfigurability in addition to being able to deal with multiple inputs ranging from light, shock, heat and even air-quality sensors to Global Positioning System (GPS) information and simultaneous monitoring of multiple radio services across a very wide frequency spectrum. This device would ideally have to have a very long working life both in terms of hardware durability and software programming. A cognitive radio may include automatic self-testing and self-healing properties to ensure the longevity and sustainability of the radio by also being able to bypass non-repairable parts of the radio if damaged. The power supply for such a device would have to have a considerable capacity, yet not result in an unwieldy device due to the size of the battery or other possible form of a portable power source. The power requirement is possibly beyond the limitations of current portable battery technology but early research work on fuel-cell and hybrid man-made and natural power source technology may increase the viability of a handheld cognitive radio.

Cognitive radio is certainly a radio technology that will prove extremely popular with both non-technically-minded users who just want to immediately use a wireless device without having to configure it themselves. Companies wishing to establish intelligent broadband communication links both in-house and between its roaming employees and the company's internal network may also find this wireless technology advantageous. Cognitive radio will most likely also become a strong generator of revenue. At this present time, the signal processing capabilities, power and physical size requirements for a feasible cognitive radio device are the main barriers to the creation of such a device.

Therefore a more achievable goal is to create a useful reconfigurable radio that will prove to be a very useful stepping-stone towards achieving the goal of cognitive radio technology and indeed, even more forward-looking wireless communication technologies.

4.3.6 Practical Reconfigurability

One desirable objective of a reconfigurable radio system is to enable a radio with a basic level of functionality to automatically increase its operational capabilities as required. This

means that a wireless device equipped with a basic communications platform installed after manufacture, can estimate what communication standards are being used on the monitored band of interest and automatically augment its inventory of modem modules. Upon estimating the channel access schemes in use therefore, the reconfigurable radio system attempts to obtain the software modules relating to the updated or new modulation and channel access schemes. If the required modules for the modulation/channel access are not in the inventory of schemes stored in the wireless device, an attempt is made to download the required software either Over The Air (OTA) download from a service provider or the closest wireless device with the required software module. If connected wirelessly or via a fixed connection to the internet, the module may be obtained via a simple software download.

4.3.6.1 Possible Future Reconfigurable Radio Scenario

The following description of a reconfigurable transceiver is one example of how a future wireless device may operate and what its capabilities may be:

Installation and activation of the module occurs without user-intervention. User activity and channel information are monitored constantly and used as inputs to the reconfiguration mechanism to maintain maximum power efficiency and maximum channel capacity usage. At the physical layer, a radio designer-specific physical layer management tool referred to in this thesis as a Spectrum Management Controller (SMC) may be used to determine the frequency spectrum access to/from the radio. This SMC controls the channel bandwidth required based on the user(s) current data transfer needs, the current channel activity, and the RF spectrum that is available. If the terminal is in a region where a spectrum usage charge is payable, the SMC requests a band segment and it is conceivable that payment of some type may be required in the future for access to the frequency spectrum.

The topic of payments for spectrum-access is beyond the scope of this thesis but, if implemented, the SMC may allow the payment layer (a processing layer in the radio protocol stack, that may be implemented at some point in the future to handle payment transactions for spectrum access) to negotiate the transaction payable by credits, tokens or some other monetary units). For high data rate, and wide bandwidth applications such as DAB and DVB-T, the SMC attempts to secure a sufficiently large segment of the available bandwidth if data transmission is required. This process involves monitoring the band of

interest to gauge the level of activity present in order to avoid busy and restricted frequencies when allocating the sub-carriers for data transmission. This vision is based on the assumption that analogue front-end hardware will evolve sufficiently so that a handheld wideband reconfigurable processor is possible. At present, large segments of the RF spectrum are not in continuous use and the possibility of renting spectrum segments to other users during these quiet periods is one idea garnering approval. With relatively short range handheld communication terminals, two or more transmissions can occur using the same frequency if the other co-channel emissions are not 'visible' by the device. As a means to increasing the spectrum usage efficiency, future reconfigurable radios will be spatially-aware i.e. able to record their positions using the Global Positioning System (GPS) and negotiate co-channel operation if the designated user is located sufficiently distant from the location as to avoid interference. It is envisaged that the present regulations regarding spectrum-usage and 'ownership' will also evolve, permitting the trial of spectrum-sharing and spectrum-rental schemes.

An external stimulus for reconfiguration may be a user who wants to transmit a large file for the minimum cost possible. An internal stimulus for reconfiguration may be the advertised costs of using each of the detected networks and air-interfaces. The radio then selects the air-interface or network which will convey the file to its destination for the lowest cost and either reconfigures the radio to enable the transmission of the file or alternatively, supplies the user with information regarding the available networks/transmission modes and associated costs. The latter option leaves the decision on which method to use up to the user, who may then initiate a manual radio reconfiguration option to enable the transfer of the file.

4.4 Conclusion

It has been argued in this chapter that OFDM communications technology may be significantly enhanced using reconfigurable radio. One of the fundamental methods of improving OFDM suggested in this chapter is to enable the radio to adapt to the time-varying characteristics of the wireless communications channel. This means that the transmission data rate, sub-carrier frequencies and transmission patterns may change in sympathy with the wireless environment in order to maximise channel capacity-usage. Reconfigurability facilitates the implementation of an OFDM receiver capable of blind synchronisation. The advantage of employing blind synchronisation methods is that the

amounts of transmitted synchronisation and receiver control data, comprising the transmission overhead, may be reduced. Data through-put rates may therefore also be increased due to the reduced size of the transmission overhead.

As a result of the discussions of reconfigurable radio in this chapter, this thesis identifies four of the many key areas where OFDM can be improved using a reconfigurable radio design. The first OFDM enhancement, presented in Chapter 7, is a means of synchronising an OFDM receiver to the OFDM transmitter using a low complexity and low transmission overhead frame synchronisation technique. The second proposed technique, described in Chapter 8, improves the performance of OFDM in noisy and interference-prone frequency bands and lowers the Peak to Average Power Ratio (PAPR) of a transmitted OFDM signal. The third proposed OFDM improvement described in Chapter 9, enables an OFDM transmitter to minimise the possible loss of information due to sub-carriers affected by excessive interference or frequency-selective noise sources. This includes a low transmission overhead means of informing the receiver of the affected sub-carriers. The final proposed method of improving OFDM-based communications, described in Chapter 10, estimates the modulation scheme used in a received signal with an unknown modulation scheme and is designed for receiver-side implementation.

Before dealing with these OFDM enhancements however, details of the chosen reconfigurable platform, which enables the implementation of these OFDM enhancements, is presented in the following chapter. This reconfigurable radio platform forms the foundation for the rest of the work presented in this thesis.

5

RECONFIGURABLE RADIO PLATFORM

5.1 Introduction

In the last chapter the wide-ranging future potential of OFDM communication systems was discussed and it was argued that the potential could be reached through enabling technologies such as reconfigurable radio. This chapter is the second main part of this thesis. In this chapter, a reconfigurable radio design platform is presented. The platform described here is used as a basis for the experimental work and prototyping solutions for next generation OFDM communication systems detailed in subsequent chapters. The chapter begins by explaining the choice of research platform. The platform consists of a general-purpose processor software radio engine, known as IRIS (Implementing Radio in Software) and a minimal hardware RF front-end. The software system is described in Sections 5.3 to 5.5 and the hardware is detailed in Section 5.6.

It is important to point out that the aim of this chapter is to describe the reconfigurable radio platform, which is used for implementing the concepts and ideas presented in this thesis. Therefore, only the essential features and methods used for the creation of a reconfigurable OFDM transceiver will be presented. For an in-depth treatment of the chosen reconfigurable radio platform, the reader is directed to Mackenzie's doctoral thesis [Mackenzie2004].

5.2 Options for Reconfigurable Radio Platforms

Several different approaches for creating reconfigurable radios exist. These include using Application Specific Integrated Circuit (ASIC), Field Programmable Gate Array (FPGA), Digital Signal Processor (DSP), and General-Purpose Processor (GPP) platforms. Each class of signal-processor architecture has an associated level of reconfiguration ability. The emphasis of this section is on choosing a signal-processing platform that will enable a

maximally-reconfigurable radio research platform. In addition, this platform must maximise exploration and development of new wireless communication concepts and techniques.

Reconfigurability at the application level is one prevalent level of flexibility and applies to systems that can switch from one mode of operation to another e.g. a GSM cellular phone that may be changed to become a DAB receiver. This aspect of reconfigurability is possible by interchanging the software and/or hardware processing structure with the requested new application structure. This application changeover occurs within the device itself (i.e. does not require the addition or removal of hardware modules). A deeper level of reconfigurability exists where the order of software and/or hardware components within each application structure may be modified in order to create or alter the structure's function(s).

A modular structure facilitates radio component interchange or even complete replacement of one or more signal-processing components/stages. This means that the demodulator associated with a GSM cellular telephone may be replaced with the demodulator associated with a DAB receiver, if a reconfigurable radio platform was used to implement these devices, for example. This is also dependent on whether the RF hardware front-end is capable of supporting both GSM and DAB frequency bands.

Consider a traditional fixed-architecture radio system approach however, where the demodulators for both the GSM cellular telephone and the DAB receiver are implemented using ASIC design. In order to implement a GSM demodulator and a DAB demodulator, two sets of ASIC hardware is required. The reason for this is that the function of an individual ASIC design cannot be reconfigured due to the application-specific nature of the device itself.

FPGA devices have limited reconfigurability options due to the physical constraints of the number of logic arrays available. They must also be taken out of service for re-programming using a vendor-specific high-level language environment, and reconfigurability in this context refers to the modification of the interconnections between the logic array themselves. It is essentially, a hardware logic reconfiguration. A change of FPGA functionality may also impact on the neighbouring FPGA, memory, I/O, and

controller devices meaning that complex management and control systems are necessary to maintain correct operation.

The flexibility and adaptability of a communications system dramatically increases as more of the signal processing functions are consigned to the purely software domain. A purely software process does not have to re-programme hardware or hardware logic devices in order to re-order or replace the signal chain or radio application. This is not the case with ASIC or single FPGA radio implementations.

Current development environments for FPGA, DSP and GPP environments offer high-level language programming methods. In the case of an FPGA system however, the programming method is based on programmable hardware logic using a Hardware Description Language (HDL) such as Verilog, or VHDL. These are also hardware-specific and simply offer high-level constructs used for expressing hardware functionality.

DSP and GPP software offer almost total reconfiguration as the desired functions are purely a set of instructions that are executed by the processor. As a result, DSP and GPP devices are more suitable for reconfigurable communication designs. DSP devices have an upper limit on the amount of reconfigurability that is possible, however. These devices rely on extra hardware support modules, memory access, controller and storage devices, and I/O controller interfaces. Management functions are required in order to maintain real-time operation, and optimise power efficiency. The reconfigurability constraints associated with DSP technology are due to the low-level bus manipulation and memory cache access routines that firstly, are critical for correct operation of the processor, and secondly, may be affected by significant changes of the DSP software processes leading to a possible failure.

Dedicated DSPs such as the Texas Instruments [TI] TMS320C6201 and TMS320C5204 are programmable signal processing machines that process signal data in the digital domain. DSPs perform single or multiple mathematical operations on the input data. Common algorithms employed for radio purposes include decimation to reduce the signal sampling rate, filtering, interpolation, differentiation, integration and multiplication. These common operations are part of the modulation and demodulation stages of a radio system. In a traditional fully-hardware radio system, these operations are implemented using a combination of common circuit components (e.g. resistors, capacitors, inductors)

encapsulated either separately, or combined as Integrated Circuit (IC) devices. These component blocks are then wired in a particular way to perform the desired function(s). In the software domain, however, the circuit function is not fixed and may be changed by simply changing the software programming code itself. This particular software radio system may more accurately be called a flexible-architecture system as opposed to the hardware-only, fixed-architecture radio.

DSP applications demand a lot of processing power due to the very high sampling rates involved. A high data bandwidth is required in order to process one or more data channels simultaneously without aliasing effects or other serious degradation of the desired signal. The device must process the data in real-time. This means that the DSP application must complete its processing tasks in time for the arrival of the next data payload to ensure reliable communications. Failure to meet a 'hard real-time' objective may result in a failure or malfunction in the application. Fixed-point signal processors truncate precision of each signal sample, warranting close attention to how this may subtly affect the processed signal. Following the digital to analogue process, the digitised signal may have a precision of say, 14 bits. A 12-bit DSP processing this data would truncate each sample to a 12-bit version of the original signal resulting in an increase in the signal to noise ratio (SNR) due to the truncation of the signal precision. Specialised interfaces and peripheral devices are required for DSPs in order to allow data input/output (I/O) and device control from a host processor. These extra interfaces and peripheral devices therefore require extra hardware and complexity in the form of memory management controllers.

The performance of DSP processes is not dependent on component tolerances or mismatches. An analogue circuit is affected by temperature changes resulting in a change of component resistance. A DSP system offers the distinct advantage of being able to perform consistently in a temperature-varying environment. Compared to an analogue circuit, the manufacturing complexity of a comparable DSP system is greatly reduced due to a modular production process. Following the DSP device creation and testing processes, the required functions are simply programmed onto the DSP. By using a generic DSP, it means that the functionality of the finished device is determined by the software itself. This eliminates the need for long application-specific production lines as the complex application is now in software. This approach allows easier testing and modification and the resulting system offers consistent function performance over a wide-ranging set of environmental factors.

As the functionality of the device is now dictated by software, the overall physical size of the system is fixed. A complex processing system that would require extensive physical circuit space if implemented using traditional analogue means, and incur high component cost and manufacturing times may be implemented digitally on a DSP for a lot less expense and reduced labour times.

Analogue electronic design tends to be much more difficult than the implementation of a signal processing solution on a DSP. In recent years, DSP versions of some analogue signal processing applications has become a cheaper and more reliable alternative. In addition to the added advantages of being smaller and offering far greater noise immunity, the testing, upgrading, and recycling processes are faster and less complex than when dealing with analogue circuits.

GPP software development is not as closely bound to the associated hardware unlike DSP designs. An abstraction layer, which shields the radio user/designer from the memory management, access, and I/O control mechanisms aids the rapid facilitation of radio reconfigurability options. The Operating System (OS) takes care of hardware specific tasks such as process thread scheduling and virtual memory access and allocation in addition to co-ordinating the input/output and other peripheral devices. Software development is therefore simplified greatly and multiple software radio implementations can run simultaneously due to the multi-threaded GPP environment. A GPP also has large amounts of Random Access Memory (RAM) allowing the storage of radio implementations for extremely fast application and/or radio structure changeover without the need for the *go offline, re-programme, and return to on-line* cycle required for FPGA and many dedicated DSP implementations. RAM and non-volatile memory permit the storage of multiple parameter sets for all the radio applications that may be required, storage of look-up tables for NCOs, and even historical samples of intercepted or generated signals for use as references.

The work presented in this thesis relies on being able to fully reconfigure the functionality of radio and air-interface. This type of reconfigurability is a mix of application, structural, and parametric control, leaving the OS to perform the essential memory management, thread scheduling, and other radio ‘house-keeping’ functions.

5.2.1 General Purpose Processors

The alternative option presented in this thesis is to exploit the signal processing abilities of General-Purpose Processors (GPP) in order to design and implement reconfigurable software, or flexible architecture transceiver techniques and applications. General Purpose Processors are more commonly identified as the standard processors in Personal Computers (PC), and laptops. The main groups are RISC (Reduced Instruction Set Computers) such as the PowerPC processor and the Intel Pentium processor range. The scope of this work is to enable extremely wide digital bandwidth access to multiple software transceivers operating simultaneously if required, but using only one hardware RF interface. A GPP forms the software processing core of such adaptable entities.

Four major classes of GPPS exist, representing the variety of approaches that processor vendors are taking to create processing devices with DSP-like performance.

- Standard GPPs.
- High-Performance GPPs.
- GPPs with built-in DSP features.
- DSP co-processors.

Moderate performance-standard GPPs, such as the Advanced RISC Machines ARM7TDMI [ARM], do not have any specific DSP features and are not suitable for intensive signal processing operations due to its poor multiplication throughput, an overhead due to the need to dynamically generate the address locations, and limited memory bandwidth.

The following sub-sections examine some of the attractive attributes afforded by GPP-based radio designs. The advantages of using a GPP machine such as the Intel Pentium processor compared to ASIC, FPGA and many dedicated DSP implementations are that extensive amounts of volatile and non-volatile memory are available for use, high-level language development environments reduce the complexity of coding required, the time to create, test, and deploy a radio, and that processor speeds are continually increasing.

5.2.2 Advantages and Tradeoffs

The numerous advantages of using a GPP show that an extremely flexible and cost-effective solution for software radio applications is possible using this platform. The use of such GPP devices was not embraced fully due to a number of disadvantages, however. GPPs can consume more power than comparable ASIC, FPGA, and DSP implementations. For power-critical applications e.g. handheld or small portable man-packs, the trade-off was reducing the functional flexibility of the radio in favour of increased battery lifetimes using these other devices.

The time taken to process the same number of data samples over a set number of test instances varies if a GPP system is used due to the overheads of thread schedulers, memory access controller functions and other processes that may also be executing on the device. The time taken to execute the same algorithm and same data source remains constant if a dedicated DSP is used as it does not have these extra overheads. As a result, the extra objective in GPP-software radio is to ensure that there is sufficient spare processing time to maintain real-time operation. Multiplexed channel access schemes have an inherent requirement for exact timing; therefore this timing jitter may result in performance degradation. GPP systems work in millisecond time increments but microsecond timing is required for some time-division multiple access schemes.

The combination of DSP techniques and control software allows the user or the device to manipulate the software parameters. This feature facilitates the creation of reconfigurable or dynamic devices, designed to adapt to the radio environment and/or the user's requirements. Reconfigurability is not exclusive to the physical layer (PHY), but refers to the reconfigurable aspects of the higher level applications using the flexible-architecture radio system also. An embedded DSP algorithm on a semiconductor device cannot be modified, reconfigured or even replaced in most cases. Using a general-purpose processor as the target device however, permits the designer to modify, upgrade, or even replace the DSP algorithm, resulting in a less expensive and a more rapid means of creating and deploying new signal-processing applications. This approach is in wide use at the present time in the industrial and consumer electronics products market. The technology has progressed sufficiently enough that a similar approach can be applied to transceiver systems. Rather than being forced to design, test and implement a new radio design using the traditional analogue hardware means, it is possible to complete the same tasks in software and prototype a new radio design without a huge manufacturing outlay

expenditure. Reconfigurability through software brings a dramatic change to radio system design, test, prototyping and deployment.

A GPP platform offers the radio designer the freedom to experiment with new wireless communication ideas and to improve, expand upon and test existing signal-processing algorithms. Instead of focussing on creating a tailored signal-processing environment for one specific task, the main focus of attention and energy is maintained on developing innovative, creative and efficient reconfigurable radio techniques.

5.2.2.1 *Memory*

Memory is an essential requirement in any digital radio design. A software-radio stores an inventory of signal-processing components in non-volatile memory and a radio controller requires memory to structure and coordinate radio applications. The outgoing/incoming signal sequences require significant amounts of temporary memory in order to be manipulated and conveyed to/from the RF front-end. RAM (Random Access Memory) is present in large quantities on a GPP platform. Of the order of hundreds of megabytes, this allows software radio applications to store sampled data and frequency synthesiser look-up tables used for the modem functions in addition to the operating radio executable code itself. A history of the signal transmission and interception is now possible for future comparison or re-transmission purposes. More interestingly, current transceiver operating parameters and the current status of the communication channels may be stored and analysed over time leading to feasible spectrum monitoring and ‘intelligent’ radios that reconfigure themselves to suit the changing channel conditions.

One other major difference between ASIC (Application Specific Integrated Circuit), FPGA (Field Programmable Gate Array), Dedicated DSP based transceivers, and GPP devices is that non-volatile memory on the scale of gigabytes can be utilised. As this memory is not normally available to non-GPP transceivers, long-term storage of sampled signal data is possible.

5.2.2.2 *Application Development Environment*

A high-level language development environment used for software radio design and implementation reduces the time required for transceiver coding, testing, and deployment

time. Hand-coded assembly language still offers the fastest algorithm execution time but at a cost of resulting in a user-unfriendly MMI, very limited scope for reconfigurability, and relatively extreme amounts of complex programming time and effort. Processor speeds have increased sufficiently enough to make real-time high-level language based software processing functions feasible however.

Unlike dedicated DSPs, a GPP platform is managed by an Operating System (OS), which takes care of the GPP management and control functions such as memory management, thread scheduling, and device controllers. The designer's time may be concentrated solely on the important algorithm development and radio topology coding. A Real-Time Operating System (RTOS) controlling a GPP radio means that DSP-like operation (in terms of attempting to maintain real-time constraints) is possible. Depending on the complexity of the radio application and the processor speed however, a non-RTOS may also function as an adequate 'soft' real-time operating system. GPPs offer one other significant advantage over other signal-processing environments; the ability of the radio designer to create and develop new wireless communication concepts and techniques is greatly improved.

5.3 Implementing Radio in Software (IRIS)

5.3.1 Overview

Reconfigurable radio represents a significant change in the concept of radio systems. This is especially apparent when compared to what the vast majority of people think of what a radio or wireless device means to them. The radical new ideas for how a radio may be designed and adapt to the wireless channel represents a change in direction from the typical all-hardware radio infrastructure. A reconfigurable radio is capable of changing the nature and capabilities of its operation autonomously. This self-arranging radio system must therefore be capable of removing, inserting and re-arranging individual signal-processing stages within the signal chain of the software radio. Additionally, a means of modifying the parameters and operating characteristics of individual signal-processing stages is essential to being able to make reconfigurable radio a reality.

To facilitate these requirements, the Implementing Radio In Software (IRIS) system was designed by Mackenzie [Mackenzie2004] to harness the processing power and multi-threading capabilities of a general-purpose processor for the creation of a reconfigurable

radio. The concepts and implementation presented in this thesis is based on using IRIS to create a reconfigurable transceiver, with an emphasis on implementing an enhanced OFDM transceiver based on a reconfigurable radio platform. In order to use this system, radio modules (signal-processing stages) must be created using Object Oriented Design (OOD) principles realised by an efficient and powerful high-level language programming language such as C++. After the creation of each module using this approach, the IRIS system may be used to structure and manage these signal-processing modules to form a reconfigurable radio. The software structure and management features used in IRIS are novel and sophisticated, thus warranting descriptions of their objectives, methodology and functions before progressing onto an example implementation.

5.3.2 Design Concepts

As a typical fixed architecture (all-hardware) radio may be composed of individual signal-processing stages, IRIS allows software versions of these stages to be called into action as Components. IRIS is designed to operate using a Windows™ Operating System using either a command line or Graphical User Interface (GUI) for overall control and visualisation of how the particular radio implementation may be constructed. Each signal-processing stage is created externally using a high-level language such as C++ and then compiled and built as a special type of software object which may be dynamically inserted or removed from the radio application while it is in operation. As a GPP platform is used, the Operating System (OS) may not therefore be a Real Time Operating System (RTOS) hence careful algorithm and reconfigurable radio platform design is required to ensure that the reconfigurable radio software processes will not exceed the real-time boundary for that particular application.

From this point on, a signal-processing stage, or module, will be referred to as an 'IRIS component', a 'reconfigurable radio component', or simply in capitalised form as 'Radio Component', 'IRIS Component', or simply 'Component'. Capitalised terms will be used from this point on to denote entities pertaining to the IRIS system.

Each signal-processing entity, or Radio Component, is constructed using a generic framework that enables every Component to connect to any or all other Radio Components as well as methods of controlling the parameters of each component. These Radio Components may be connected in series or parallel using the IRIS system to control how

the flow of information between the Radio Components. Initial radio configurations are specified using a configuration file, which is an intuitive method of arranging the components in a either a series or parallel structure, or even a combination of series and parallel structures that defines the specific radio implementation in mind. These configurations may be stored, amended and used multiple times as required. An inventory of Radio Components may be created by the radio designer or obtained from other sources that use the IRIS system. Using this inventory of components, configuration files may be used to arrange a chosen number of these components to form a reconfigurable radio. This enables the designer to rapidly deploy, or switch to a different application by choosing a new configuration file.

The IRIS architecture may be used to implement a standard software radio (i.e. a software radio which does not use reconfigurability) as well as enhanced reconfigurable radio designs. A generic configuration mechanism and the same constructs and design principles may be used for all software radio/reconfigurable radio designs.

5.4 IRIS Software Architecture

5.4.1 Radio Framework Outline

IRIS is based on a framework that uses common constructs and design methodologies for every IRIS radio application. Each Radio Component is built using a generic structure, comprised of a connection interface and controlling mechanism common to all other Components. This enables each Component to be connected to every other component if required. For example, this means that the implementation of say, a low-pass filter, the creation process is the same as for an OFDM transceiver. This IRIS architecture, or the prevailing superset of principles in the IRIS system design defined by Mackenzie, comprises a Component Framework, a component model and a defined set of rules for the creation of a radio and component control and management system called Control Logic [Mackenzie2004].

5.5 Main Framework Entities

The Radio Framework is based on five major entities, as shown in Figure 5.1:

- Radio Component.

A Radio Component is the basic unit of an IRIS implementation and comprises an individual stage in the signal-processing chain of an IRIS reconfigurable radio/general signal-processing application. The actual level of functional complexity that a Radio Component may encapsulate is at the discretion of the radio designer.

- XML Parser

An XML parser is used to interpret the radio design specified in the XML configuration file and convert this information into a form that may be processed by the Component Manager and Radio Engine

- Component Manager

The Component Manager loads the Radio Components specified in the XML configuration file and unloads a previously loaded radio configuration. The Component Manager also compiles an inventory of Components which may be located either on the host PC or from a remote location connected via Ethernet or internet, and these comprise the available Components that may be used as part of a radio implementation.

- Radio Engine

The Radio Engine is the core of the IRIS framework. This implements the radio design to create an actual reconfigurable radio using the IRIS framework as the management and controlling entity. The Radio Engine relies on the Component Manager and control interface known as Control Logic.

- Control Logic

IRIS uses a Control Logic Manager (CLM) which is the main means of reconfiguring a radio configuration when the radio is in operation. This radio management feature can access the structure and parameters of all the Components and modify them as required. The CLM is independent of the Components (i.e. the CLM is a separate process that connects to the radio using a common interface and is fixed by any particular Component) and therefore may externally modify the operation and parameters of any of the Components that comprise the radio implementation. Control Logic will be explained in greater depth later in this chapter.

- IRIS Application Programming Interface (API)

The API used in IRIS is used to integrate an IRIS reconfigurable radio application into other applications. Specific functions that manipulate and control the IRIS Framework are abstracted from the core of the IRIS system enabling the sophisticated IRIS features to be accessed using a simple programming interface.

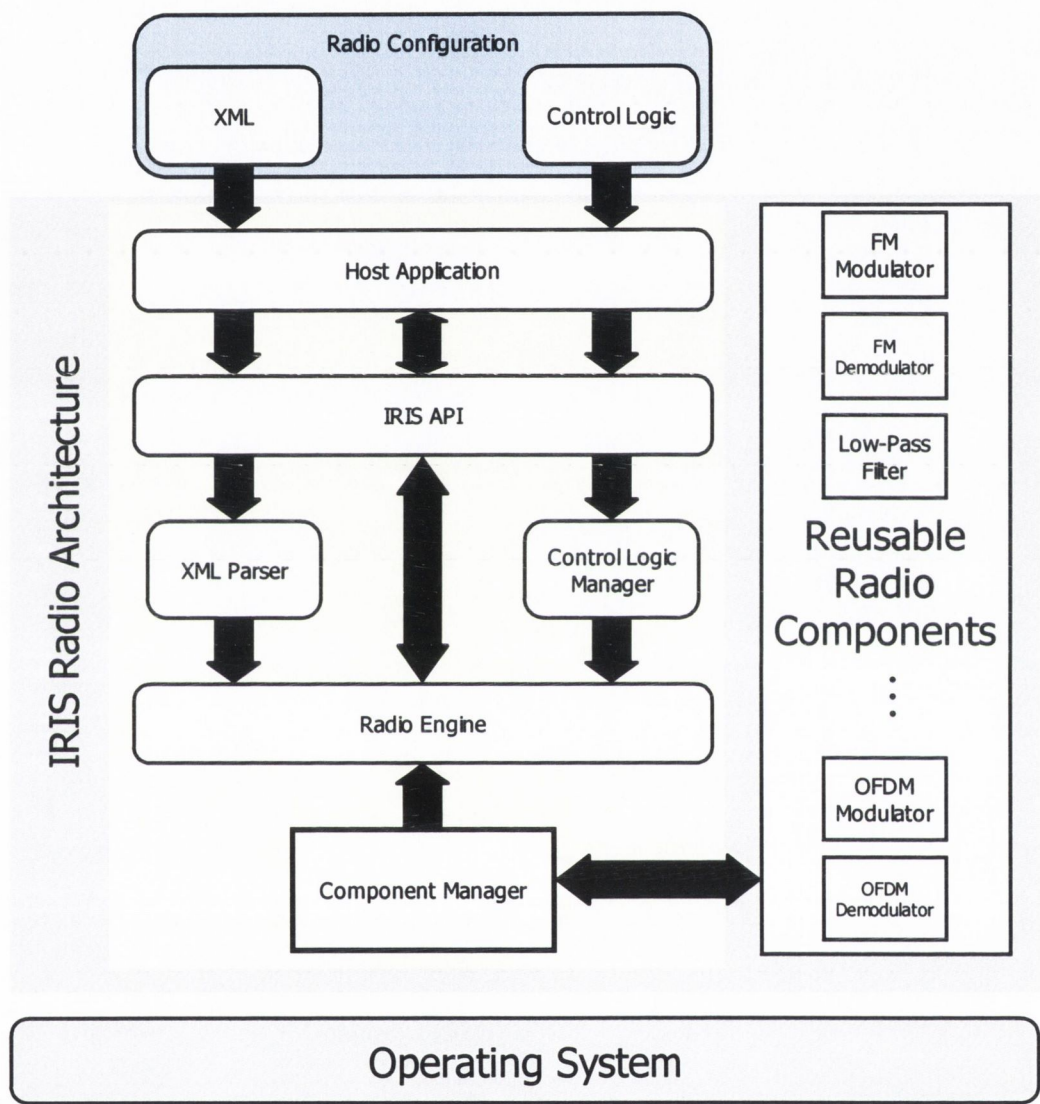


Figure 5.1: The IRIS Radio Architecture

Each of these main parts of IRIS, outlined above and illustrated in Figure 5.1, will be examined in more detail as they are essential to the creation of a reconfigurable radio and in particular, the creation of a reconfigurable OFDM transceiver as described in this thesis.

5.5.1 Radio Component

An IRIS radio implementation is created using a sequence of Radio Components. Each Radio Component may encapsulate one or more signal-processing functions with generic

methods of interfacing with any other Radio Component and the rest of the IRIS Framework. A Radio Component may be used in multiple instances without having to create a new version of the particular Radio Component. The approach leads to an efficient and manageable reconfigurable radio design.

Discussed in the previous chapter was the point that a reconfigurable radio designed for prospective wireless devices must be capable of application, structural and parameter-level reconfiguration. In order to achieve these goals, the first step of the process is to enable reconfigurability of the fundamental unit of the IRIS framework, namely the Radio Component. Mackenzie's original design requirement was to create a Radio Component structure that would facilitate maximum flexibility in terms of access to the parameters and structure of each Radio Component in order to modify its' behaviour and operational characteristics even while the Radio Component is in active use in an IRIS reconfigurable radio application.

The level of reconfigurability is referred to as the 'granularity' of the Radio Component. Application-level reconfigurability is deemed to be 'large-grain' reconfigurability; structural reconfigurability with each application may be referred to as 'medium-grain' reconfigurability, or if the parameters of the Radio Component itself may be modified, this may be referred to as 'small-grain' reconfigurability.

In order to reduce the level of complexity that a user may be subjected to when dealing with the IRIS Radio Components, a layer of abstraction is used to reduce the amount of unnecessary visibility into the internal structure of each Radio Component and provide a more straightforward method of interfacing with the IRIS Framework and other Radio Components. In essence, each Radio Component is designed using a 'black box' concept, where the designer may decide to expose only the functions and parameters that are deemed necessary and useful to the user by means of standardised interfaces.

Each Radio Component may trigger one or more 'events', which occur when a set of criteria defined by the designer are fulfilled. This enables other processes within the radio implementation to respond to the triggered event. This feature is very useful for activating a reconfigurable radio which has gone into sleep mode when a signal of interest has been detected, for example. A more common example of where events may be used is where a process needs to notify a data source that the input signal has been processed and new input data is required.

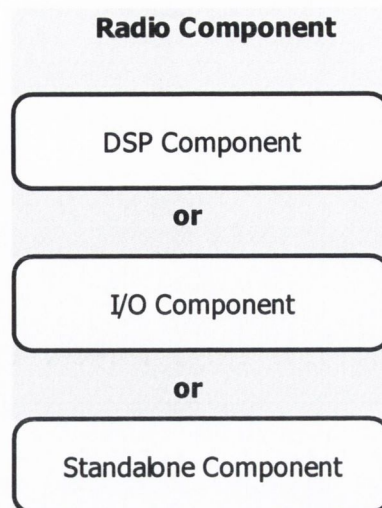


Figure 5.2: Radio Component types specified in the IRIS Framework

Three different Radio Component types may be created using the IRIS system and each of these component types represents a different layer of abstraction from the IRIS core framework. A Radio Component may either be a DSP component, I/O component or a Standalone component. An Input/Output (I/O) Radio Component is specified in the IRIS Framework for dealing with I/O routines that either act as a means of conveying information to/from a sink/source such as a file, audio device or an Internet Protocol (IP) interface. These I/O Radio Components therefore have extra constructs for enabling the supply or delivery of information to/from the rest of the signal-processing chain specified in the radio configuration description in addition to the constructs required for the normal signal-processing routines encapsulated in the Radio Component. DSP Radio Components that deal specifically with DSP routines comprise the majority of the possible Radio Components may be used as part of a typical reconfigurable radio implementation such as a reconfigurable OFDM transceiver. These DSP Components do not have I/O-specific constructs as the source information is expected to arrive from the previous Radio Component in the signal-processing chain and the destination for the processed information is expected to be either another DSP Component or an I/O Component, such as an I/O Component that stores the processed information in a file. The third type of Radio Component is called a Standalone Component. This is similar to a DSP and I/O Component in terms of encapsulated signal-processing routines but the difference is that this Radio Component type contains constructs for extra functionality that is not part of the signal-processing chain such as access to an external hardware device or external timers. The component type may be used to change the band of interest that is being accessed by a

hardware RF front-end or trigger external events such as activating a second RF front-end in order to simultaneously process signals from more than one frequency band.

5.5.2 XML Parser

An IRIS-specific XML configuration is used to define a radio application or even a general signal-processing application. This is the first stage of the reconfigurable radio creation procedure. XML was chosen by Mackenzie as it facilitates hierarchical description of the radio entities in addition to intuitive and relatively uncomplicated design process. The XML configuration lists the Radio Components and the initial parameters each Component possesses. The Radio Components may be arranged in series or parallel, or even using a combination of the both. The design methodology is intuitive as it is based on a ‘block-diagram’ approach where each Radio Component may be placed in the configuration file that corresponds to the expected signal flow through the radio/general signal-processing application.

The example shown in Figure 5.3 is also useful for describing how the structure of a radio may be defined. The order in which the Radio Components appear follows the expected signal flow of the application. The XML configuration for a reconfigurable radio design or a general signal-processing application must be encapsulated with the `<reconfigurableradio>` and `</reconfigurableradio>` XML tags before inserting the description of the required Components between the `<structure name="structurename">` and `</structure>` tags. This represents the ‘packaging’ of the reconfigurable radio/general signal-processing application and this template is required for

```
<reconfigurableradio>

    <description>
        <name>IRIS Application Name</name>
        <comment>Comment</comment>
    </description>

    <structure name="structurename">

        // IRIS Components here

    </structure>

</reconfigurableradio>
```

Figure 5.3: Basic IRIS reconfigurable radio/signal-processor XML configuration.

all such designs. The XML configuration file may specify one or more radio implementations comprising single or multiple signal chains in parallel.

The XML Parser is used to translate the XML configuration information, which has the basic structure as shown in Figure 5.3, into an actual reconfigurable radio/general signal-processing application. The first stage of this process is to verify that the XML configuration is correct and then convert the Radio Component hierarchy into a suitable representation that the Radio Component Manager may then obtain the IRIS Components from the inventory or available components and arrange them in the required format. Specifically this means that the main work of the XML parser is due to translating the XML Radio Component arrangement specified between the `<structure name="structurename">` and `</structure>` XML tags into a chain of signal-processing stages in software.

5.5.3 Radio Component Lifecycle

Each Radio Component undergoes a specific lifecycle, or chain of events that define its pattern of operation. Before concentrating on a specific example of a reconfigurable radio application such as an enhanced OFDM transceiver, it is important to describe the underlying processes and design criteria for creating such an application. This section focuses on the general lifecycle of each Radio Component, which is composed of five major stages.

- Component Loading
- Initialisation
- Commencement and Data Processing
- Halting
- Cleanup and Component Unloading

5.5.3.1 Component Loading

The Component Loading stage prepares the Radio Component for use within the IRIS reconfigurable radio application. This involves firstly obtaining the component (in the form of a compiled and built software entity) from the inventory of available components and then creating an instance of the class that defines it. Class instantiation is required so

as to be able to employ the functions encapsulated in the class declaration, in the reconfigurable radio application.

5.5.3.2 Initialisation

The initialisation stage prepares the instance of the Radio Component for use. The XML configuration description of the Radio Component specifies the parameters and input/output data types that will be used in each component instance. The initialisation stage uses the information given by this XML description to allocate sufficient memory for these parameters, prime them with initial values and establish the input/output data types that will be used to receive and pass on processed information to/from other Radio Components in the signal chain. Each XML radio component description corresponds to the parameters used for one component instance. Parameters and initialisation data are not transferable to any other instance of the same component when multiple instances of the same component are created. In this case, separate XML component descriptions must be provided for each additional component instance.

5.5.3.3 Commencement and Data Processing

This stage is the start-up phase of the Radio Component lifecycle where the incoming information is processed. If the Radio Component is an I/O component, then this phase commences the flow of information from the input device or source (e.g. a file, audio source, IP network traffic) and/or opens the destination for writing processed information (e.g. an output file, speakers, IP network). If the Radio Component is specified in the XML configuration as a DSP component, the Radio Engine calls the main processing function named `Process()`, which is defined by the designer and contains the signal-processing algorithm(s) specific to that particular Radio Component. The input to each Radio Component instance is information relating to the memory address of the input data (i.e. the memory address of the start of the block of data in the programme memory) and the data type associated with this data. This simplifies the radio processing signal-chain, and improves the efficiency of the Radio Engine and Radio Component programming as the sequence of data does not necessarily have to be copied from one memory location to another. Each component processes information when requested to do so by the Radio Engine. These components are therefore 'passive' resulting in far less radio management complexity due to the asynchronous operational nature of each component, which eliminates the requirement for a means of synchronising the possibly many different

processes that comprise the reconfigurable radio. The `Process()` function is called repeatedly by the Radio Engine when a new input data sequence is ready to be processed.

An IRIS I/O component may invoke designer-specified `Start()` and `Stop()` functions in addition to the `Process()` method as the intended use of an I/O component is to convey information to/from an external resource such as a file or ADC. The `Start()` method begins the flow of information from the resource and when `Stop()` is invoked, its intended objective is to halt the input and/or output data stream(s).

IRIS DSP components differ from I/O components as they actually have two different variations of the `Process()` method. One variation of this method allows the data located in one memory location to be processed and then the result is stored in the same memory location as the original. In effect the original data value is replaced with the processed data value. This procedure is called ‘in-place’ signal-processing and the main advantage is that extra memory does not have to be allocated for storing the results of the `Process()` function thus preserving the memory consumed for the reconfigurable radio application and minimising the amount of processing power that is required to allocate extra memory in the first place. In-place processing only applies to signal-processing functions where the size of the processed data is the same as the input data (i.e. the memory size of the input and output data must be the same). If this is not the case, then the second variation of the `Process()` method must be used. This particular ‘not-in-place’ `Process()` function variation uses separate memory locations for the input and output (processed) data sequences. It is important to note that the inputs to each variation of the `Process()` method are the memory address of the data and not the data itself. This feature significantly reduces the amount of memory allocation, copying and de-allocation as only the minimum amount of memory required for the application may be allocated. As a result, this means that the majority of the processing power is preserved for the vital main signal-processing routines and IRIS management and control entities.

5.5.3.4 Halting

This part of the component lifecycle is only relevant to I/O components and stops the I/O transfer. As a result, if the I/O component was obtaining data from the source, then the rest of the signal-processing chain would also halt as no new data sequences would be presented for processing using the `Process()` function.

5.5.3.5 Cleanup and Component Unloading

The clean-up procedure involves invoking a function called `Destroy()`, which closes any opened external resources such as source or sink files and/or I/O devices. This function also de-allocates temporary memory that was used for computing partial results of the signal-processing functions used in the Radio Component.

When all of the signal-processing functions have halted, memory used for the programme operation has been released back to the Operating System (OS) and any external resources have been detached safely. The last stage of the Radio Component lifecycle is to unload the instances of all the Radio Components used in the reconfigurable radio application. This simply involves deleting the instance of the Radio Component.

5.5.4 IRIS Lifecycle Outline

In order to provide a clearer picture of how the individual stages described above may form a cohesive reconfigurable radio device, this sub-section summarises the pattern of operation for an IRIS reconfigurable radio from the initialisation to shutdown stages of operation.

The structure, parameters and external control entities of a reconfigurable radio implementation using the IRIS system may be described using an XML configuration file. The XML parser may then traverse this configuration file and verify that the description syntax is correct. After the parser and scripting engine has translated the XML description into an IRIS programming code version, the Radio Engine employs the Component Manager to locate and load instances of the desired Radio Components in a structure according to the specified reconfigurable radio design. If an application-specific Control Logic has been specified, the Control Logic Manager locates and loads an instance of this external controller algorithm. Resource allocation, component initialisation and radio verification comprise the concluding step of the initialisation process.

During the operation of the IRIS reconfigurable radio, the Radio Engine maintains data flow by invoking the processing functions encapsulated in each of the active components and routing input and output information to/from components. If user-defined Control Logic is involved in the radio application, this manages and dynamically reconfigures the radio using the Control Logic interface. Upon entering the radio shutdown period, the

Radio Engine is halted. Resources such as memory, external file sources/sinks and any attached external devices such as ADC and DACs are released and halted. This is achieved by invoking the individual clean-up functions contained in each component before the instance(s) of each component is/are deleted.

5.5.5 Component Interfaces

This section examines the interfaces which are used to interact with the Radio Components and the IRIS framework that are abstracted from the base-level programming structure of the IRIS framework entities. An interface layer is a higher level set of functions that enable a device/ Radio Component to be manipulated without having to handle the low-level and much more complicated software programming code of the device/Radio Component itself. Several IRIS interfaces have been created in order to allow the designer and indeed, the IRIS framework itself to interact with other IRIS entities using well-defined and straightforward methods. The enhanced OFDM reconfigurable transceiver described layer in this thesis relies on these interfaces to control and modify the operation of the IRIS radio system therefore a description of the properties and the capabilities of these interfaces is necessary. A more in-depth explanation of the available functionality for each interface may be found in Mackenzie’s doctoral thesis [Mackenzie2004].

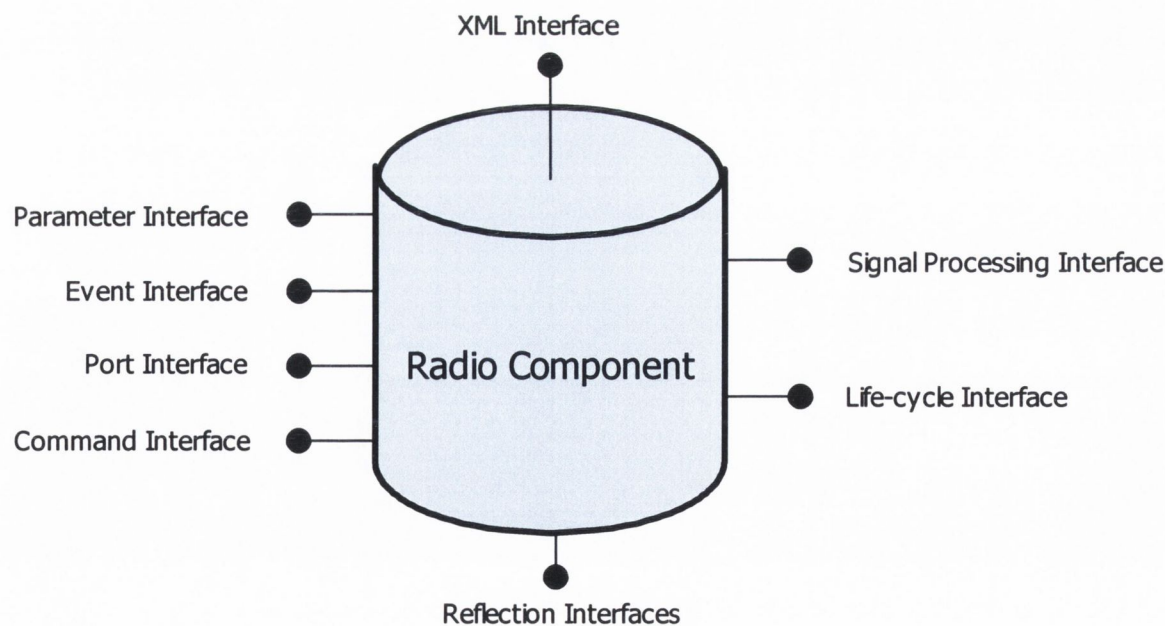


Figure 5.4: Available Radio Component interfaces.

Figure 5.4 is an illustration of how the internal operation and parameters of each Radio Component may be accessed by the IRIS framework entities. The internal software

programming constructs and structure may only be accessed using this set of interfaces resulting in a less complex and more easily manageable radio system. It is important to note that every Radio Component used in the IRIS system may be accessed using the same interfaces. When considering that a reconfigurable radio implementation may contain a significant number of Radio Components, system design and control are therefore far less complicated processes than the alternative of having to deal with Radio Component-specific handling routines.

5.5.5.1 *Life-Cycle Interface*

The Radio Component lifecycle was discussed previously in Section 5.5.3 and the functions associated with controlling this life-cycle form the Life-Cycle Interface, which is one of the interfaces illustrated in Figure 5.4. It was also shown that DSP, I/O and Standalone Radio Components have slightly different life-cycles and these are accommodated for, using this interface.

5.5.5.2 *Reflection Interfaces*

The Reflection Interfaces allow other IRIS framework entities to obtain information regarding the structure and input/output data types used in a particular Radio Component. These interfaces may be used with the XML interfaces in order to extract or set parameters or other information in the XML configuration file. The types of parameters and commands that are used in the component, which may be accessed by other Radio Components through the Control Logic interface, may be found by using a Parameter and Command Interface. Each component may be designed to asynchronously trigger 'events' in order to control an external device or influence the operation of one or more Radio Components in the radio implementation using Control Logic. Apart from general information such as the author and version of a particular component, external entities may also query the Radio Component regarding what ports (if any) have been integrated into the Radio Component to allow asynchronous transfer of information to each Radio Component, using Control Logic to manage this alternative means of influencing the processing structures within each component.

5.5.5.3 *Parameter Interface*

The parameter interface is one of the most important interfaces as it enables the radio designer to gain access to the parameters used in all of the components. This is required in order to reconfigure the Radio Component as well as monitoring the current operating parameters. Two functions comprise this interface; one function obtains the parameter value associated with a particular parameter name and the second function may be used to set the value of a particular parameter. In a similar fashion to the rest of the interfaces, the Control Logic implementation may be used to extract and set parameter details from any of the Radio Components used in the associated IRIS XML configuration.

5.5.5.4 *Event Interface*

The IRIS framework also incorporates a means of informing external radio clients or other signal-processing entities not necessarily part of the IRIS framework that some criterion has been satisfied. This is called an event and information regarding events is handled using the Event Interface. Radio Components may trigger events asynchronously to notify other Radio Components and external devices of occurrences in the reconfigurable radio operating lifetime. Information may also be sent to the entity that is listening for an event and there are no limits to the number of events that may be set by each Radio Component. A conceptual example of where events may be used is a reconfigurable OFDM transceiver which may use events and Control Logic to increase or decrease the number of sub-carriers used in the transmitter based on information about the spectrum-activity as measured by the receiver.

5.5.5.5 *Port Interface*

Ports act as inputs into Radio Components and form an alternative method of passing information into the component. This information is not restricted to primary input signal sequences but may be feedback information for use within the Radio Component. For example, an Automatic Gain Control (AGC) may receive its negative feedback signal through a port, or Received Signal Strength Indicator (RSSI) information may be used to aid the modulator component to choose a more efficient or robust modulation scheme. The port interface may be used in conjunction with the event interface. A particular radio implementation may trigger an event to signify that the information received from a port has been processed.

5.5.5.6 Command Interface

It may be necessary to invoke designer-specific or in-built functions within a Radio Component in order to reset, enable or disable the component, or carry out other specific tasks. The Command Interface enables these functions to be invoked asynchronously using the Control Logic implementation. Examples of possible scenarios that may use the Command Interface are when a Radio Component needs to reset to its initial state or when the transmitter or receiver needs to be de-activated (either under remote or local control) in order to preserve power or prevent interference.

5.5.6 Component Manager

The IRIS Component Manager (ICM) compiles an inventory of, and creates a reconfigurable radio/general signal-processing application using the fundamental unit of the IRIS reconfigurable radio, which is the Component. The ICM relies on two main pieces of information from the XML configuration. The first of these is the list of Components which are required for the application. These Components may then be located in the inventory. The second main piece of information required for a reconfigurable radio is the order in which these Components are required to appear in the signal-processing chain.

5.5.7 IRIS Radio Engine

The Radio Engine is the most important software management process in the IRIS framework. This creation of an actual reconfigurable radio using the IRIS framework relies on the Radio Engine to convert the parsed XML configuration into the desired radio/general signal-processing application comprised of assembled and structured Radio Components. The Radio Engine is responsible for the operation of the reconfigurable radio. This involves initialising each of the Radio Components, conveying information to/from each of the Radio Components specified in the particular XML configuration and invoking the algorithms encapsulated in each of the Radio Components when data is available to be processed. In addition, the Radio Engine invokes the shutdown procedures in each of the Radio Components when it attempts to halt the entire reconfigurable radio. The shutdown procedure in each Radio Component de-allocates memory and detaches the Radio Component from peripheral devices used for the radio application in addition to

Radio Engine is that it may access and modify the internal structure and parameters of the radio design using a Control Logic interface to realise a true reconfigurable radio.

5.5.8 IRIS Graphical User Interface (GUI)

The IRIS system offers a GUI for rapid radio design and initial configuration of the components associated with the radio implementation. This GUI enables all of the available Radio Components stored in the inventory to be browsed and inserted into the design if required by the designer. An example of this user interface may be seen in Figure 5.5. The figure shows the radio description using XML and in this case, an application involving a simple signal generator and an OFDM modulator is shown. In addition to the design of a reconfigurable radio application, the GUI may be used to initiate and control a practical IRIS reconfigurable radio implementation. The main advantage of this GUI is that rapid design modifications and tests made be made and as all of the radio management and control functions are encapsulated in the IRIS system, the designer may concentrate on the functionality of each component and intended radio application.

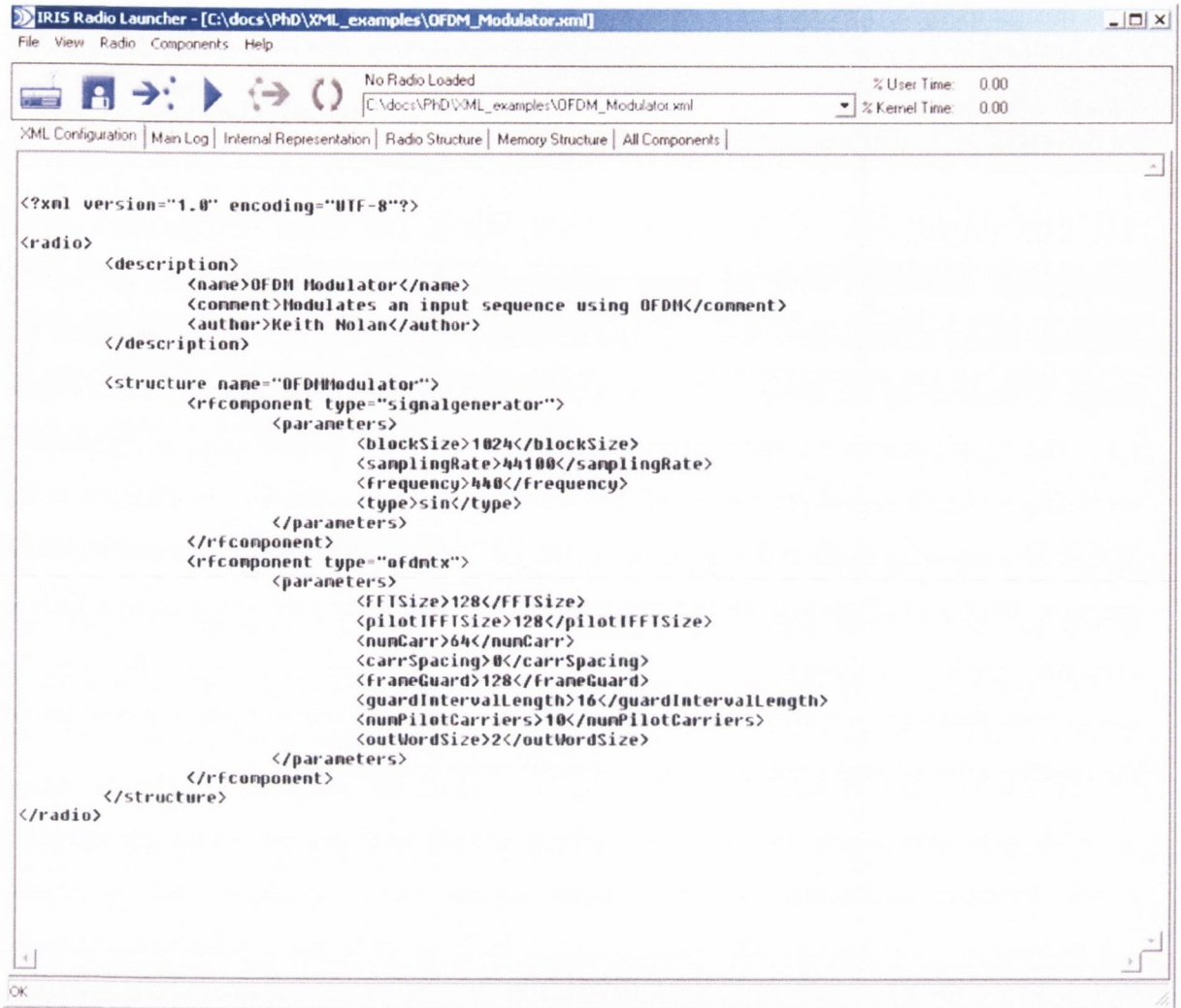


Figure 5.5: IRIS GUI radio description screen.

5.5.9 Control Logic

One of the main objectives in creating a reconfigurable radio is implementing a means of accessing the internal structure and parameters of each of the Radio Components used in each reconfigurable radio design. The IRIS system uses a concept called Control Logic, which is a means of dynamically controlling the actual radio application, radio structure and the parameters used in each Radio Component. The reason for this Control Logic is that when Radio Components are grouped together in series or parallel signal chains (or a combination of both serial and parallel chains). A number of parallel signal-processing stages or unique signal chains may operate asynchronously but a means of co-ordinating access to shared resources such as data files and external resources such as the ADC and DAC may be required. Inter-component dependencies may occur. This essentially means

that if the parameters or function of one component in the signal chain changes, it may mean that one or more of the rest of the components in the signal chain may need to update their operating parameters and/or other characteristics. For example, consider the case where one of the components in the signal chain of an IRIS reconfigurable radio is a modulator and the modulation scheme in use is BPSK, which modulates a sequence of N bits each time the modulator function is invoked. This results in a sequence of N modulated signal samples, which are complex-valued. If the device user or the radio itself then chooses a more complex modulation scheme such as QPSK, this means that the bit-to-symbol ratio changes. In this case, two input bits are mapped to one QPSK symbol. Each time this modulator is invoked, the input sequence of N bits is therefore converted to a modulated complex-valued sequence of length, $N/2$. As a result of this, the next component in the signal-processing chain must be notified that the expected length of the input signal sequence has changed to $N/2$.

This necessitates a means of changing one or more of the parameters used by one or more of the Radio Components in the reconfigurable radio design. Each Radio Component has to exist independently of all of the other components in order to facilitate dynamic component insertion and removal used for reconfigurable radio structures and applications. Therefore, each component in the radio design is not capable of obtaining information about the other components attached to it, by itself as this would mean that the component was ‘tied’ to the application. Each component only ‘sees’ an input comprising of a first memory address of the data sequence to be processed and the data type relating to the input data sequence.

In order to fulfil the objectives of maintaining component independence from any IRIS reconfigurable radio application yet facilitate component parameter and operational objective updates if one or more of the Radio Components in the signal-chain is modified, the IRIS system offers a means to control and co-ordinate this feature using a software concept called ‘Control Logic’. Control Logic comprises extra software programming that is specific to the particular IRIS reconfigurable radio application. This means that the Control Logic programming code is tied to the radio application but the Radio Components under its control are not. The Control Logic software may access one or all of the Radio Components in the IRIS reconfigurable radio design using functions abstracted from the main IRIS framework, or known as the Control Logic interface. This interface may be ‘visible’ to all of the components if required, thus allowing information regarding the

current states and parameters of all of the Radio Components in the reconfigurable radio design to pass to/from one component to one or more other components. Support for this Control Logic functionality is integrated into the IRIS Framework in the form of a Control Logic interface (or defined constructs that may be used in the IRIS Framework. This information-passing is enabled through the use of an API that allows queries and data to be passed to and from the Radio Components. The Control Logic designed for a particular application (a specific XML reconfigurable radio configuration) is application-specific but yet it is abstracted from the radio implementation structure. This means that Radio Components may be added, removed or modified in the radio implementation that is in operation and the Control Logic will still function as normal. The Control Logic feature has been designed to be user-defined using Java and C++ programming languages. For the purposes of the work presented in this thesis, C++ is used to maintain programming code consistency and ease of design.

5.5.9.1 Control Logic Lifecycle

As is the case with the Radio Components, the Control Logic has its' own lifecycle that is accessed using a simple interface that is consistent for every IRIS reconfigurable radio application. Three functions, which are shown in Figure 5.6, are required to integrate Control Logic functionality into an IRIS reconfigurable radio application and the use of Control Logic must be stated in the XML reconfigurable radio configuration. These will now be explained in turn.

```
//Invoked when the reconfigurable radio is being loaded
void Load(EngineInterface *eng);

//Invoked to initialise Control Logic
bool AttachToComponents();

//Invoked during the unloading procedure
void Unload();
```

Figure 5.6: IRIS Control Logic lifecycle functions.


```

<reconfigurableradio>

    <description>
        <name>Application Name</name>
        <comment>Comment</comment>
    </description>

    <control>
        <implementation>ControlLogic.dll</implementation>
    </control>

    <structure name="structurename">

        // IRIS Components here

    </structure>

</reconfigurableradio>

```

Figure 5.7: Basic IRIS reconfigurable radio/signal-processor XML configuration using Control Logic.

- Load()

This function is automatically invoked by the Radio Engine to initialise the Control Logic specified in the XML configuration. An example of this is shown in Figure 5.7, where the basic IRIS design template shown in Figure 5.3 has now been updated to include Control Logic. The Control Logic is specified using the `<control>` and `</control>` XML tags where the actual Control Logic implementation is a Dynamic Link Library (DLL) created using the C++ programming language.

- AttachToComponents()

Some initialisation within the Control Logic implementation itself may be required. This is designer-specific and depends on the complexity of the Control Logic implementation. This function invokes any extra initialisation functions and attempts to connect to the Radio Component parameters that will be used to influence other parameters and operating characteristics of one or more other Radio Components.

- Unload()

This is the final method in the lifecycle of the Control Logic implementation that may be invoked. As the name suggests, this function unloads the Control Logic from the reconfigurable radio implementation and release the resources which were used during its' operating lifetime (e.g. memory, external I/O devices and files).

5.5.10 IRIS Application Programming Interface (API)

The next stage is to begin the creation of a reconfigurable radio. In order to illustrate the principles of an IRIS design, a simple example is chosen. Interpolation (or Up-Sampling) is an example of a simple signal-processing application that may be used as part of a much larger reconfigurable radio application. An example of this basic application is shown in Figure 5.8. This common signal-processing operation requires an input stage, an interpolation stage and a post-interpolation filter (a low-pass filter used to minimise possible signal aliasing). This figure shows how a sine-wave signal generator, interpolation and low-pass filter Components may be connected in series using the XML configuration file. The parameters of each Radio Component are also shown in this figure. The particular parameters that may be included in each Radio Component description are at the discretion of the designer. It is possible to expose all of the variables that are used in each Radio Component in this way, allowing their values to be specified in the XML Component format shown in this figure. To reduce the complexity of each Component XML description, only the essential parameters of each Radio Component may be exposed, if required. For example, the low-pass filter XML description shown in Figure 5.8 only specifies the number of filter taps to use and the cut-off frequency of the filter, which are the most important parameters in this Radio Component. The low-pass filter Radio Component calculates the filter coefficients using optimised signal-processing functions from the Intel Signal Processing Library (SPL), which are incorporated into the initialisation code of the component. Each of the parameters described in Figure 5.8 can also be modified while the radio is in operation using the IRIS Control Logic feature to dynamically change the values of variables. When a filter value is changed, the filter initialisation procedure must be re-invoked in order to correctly implement the desired filter.


```

<reconfigurableradio>

  <description>
    <name>Interpolation Example</name>
    <author>Keith Nolan</author>
    <comment>Implements_an Interpolation procedure</comment>
  </description>

  <structure name="Interpolator">
    <rffcomponent type="signalgenerator">
      <parameters>
        <blockSize>1024</blockSize>
        <samplingRate>44100</samplingRate>
        <frequency>440</frequency>
        <type>sin</type>
      </parameters>
    </rffcomponent>
    <rffcomponent type="interpolator">
      <parameters>
        <outputBlockSize>4096</outputBlockSize>
        <filterCutoff>0.3</filterCutoff>
        <numTaps>16</numTaps>
      </parameters>
    </rffcomponent>
    <rffcomponent type="lowpassfirfilter">
      <parameters>
        <filterCutoff>0.3</filterCutoff>
        <numberTaps>16</numberTaps>
      </parameters>
    </rffcomponent>
  </structure>

</reconfigurableradio>

```

Figure 5.8: XML configuration for a signal generator and interpolator using IRIS.

5.5.11 Synchronous and Asynchronous Radio Operation

Prior to this section, the focus has mainly been on single direction signal-paths in order to preserve the clarity of explanation. Not all reconfigurable radio structures have a single signal-processing path as previously illustrated. A useful reconfigurable radio must be capable of dealing with bi-directional signal-paths as in the case of a transceiver. An IRIS reconfigurable radio implementation must therefore support modulation and demodulation in either synchronous or asynchronous modes (i.e. synchronous mode means that the transmitter and receiver are operating in the application, or the reconfigurable radio is operating as either a receiver or a transmitter, which is referred to as asynchronous mode in the thesis). This section describes the constructs which have been integrated into the IRIS system to facilitate more than one signal-processing application operating at a time.

Of particular interest to the work presented in this thesis is that IRIS has built-in support for both synchronous and asynchronous operation by enabling multiple radio structures to be defined in the XML reconfigurable radio configuration. In fact multiple input and output channels may be defined, limited only by the available processing power to process the data sequences relating to each channel. The approach means that one structure may define a receiver and a second structure may define the receiver, or multiple receivers or transmitters may also be defined. This is important because a reconfigurable radio requires at least one asynchronous receiver and one asynchronous transmitter, not only to form a transceiver, but also in order to reconfigure the radio based on the monitored spectrum activity.

A synchronised signal-path may be used to ensure that each stage of the signal-processing

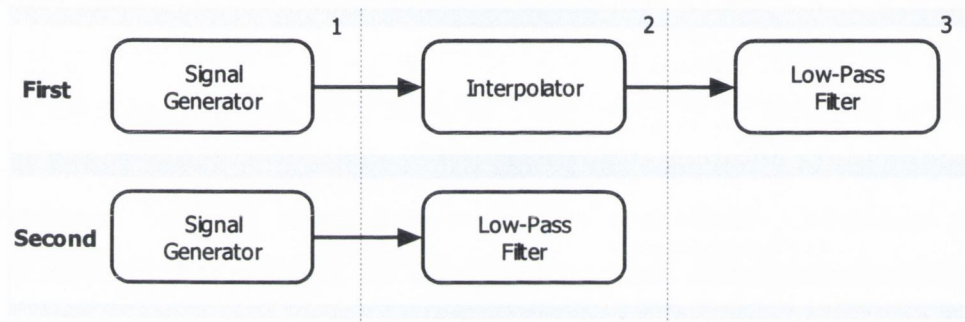


Figure 5.9: Block diagram of two parallel signal-processing applications.

chain is completed before progressing onto the next stage of that particular reconfigurable radio implementation. This is an important feature due to the fact that since the required processing time for each stage may differ, one stage may complete processing in a shorter time than one or more others. This may mean that the next stage in the signal-processing chain may be reached before all the other channels have completed the preceding signal-processing stage. Multiple structures may be defined in the XML configuration using the `<parallel>` tag where two or more reconfigurable radio structures may be defined as long as they are encapsulated in separate `<parallel>` and `</parallel>` tags. Figure 5.9 illustrates two basic signal-processing applications arranged to operate synchronously, with each operational stage marked 1 to 3. An example of the XML configuration description for these simple IRIS applications operating in parallel and synchronised is shown in Figure 5.10. The `<parallel name="first" exclusive="false">` parameter shown in the XML configuration in this figure specifies that each application is not exclusive. This means that the first interpolation example application and the second low-pass filtering application operate synchronously.

The second case where two or more IRIS applications operate in parallel is that each individual application may be configured to operate asynchronously and as two simultaneous processes (or actually, pseudo-simultaneously as they are controlled by the OS thread scheduler). Regarding the two simple signal-processing routines shown in Figure 5.9, these made be configured to operate asynchronously by changing the `exclusive="false"` to `exclusive="true"` for the both of the structures in the XML configuration description. In both cases, inter-application (or inter-structure) communication is achieved through the use of Control Logic, which operates as an external control structure that is specific to the XML configuration file as previously discussed. Examples of Control Logic will be given in subsequent chapters.

```

<reconfigurableradio>
  <description>
    <name>Interpolation and Filtering in parallel</name>
    <comment>Parallel structures example</comment>
  </description>

  <structure name="InterpolationAndFiltering">
    <parallel name="first" exclusive="false">
      <rftcomponent type="signalgenerator">
        <parameters>
          <blockSize>1024</blockSize>
          <samplingRate>44100</samplingRate>
          <frequency>440</frequency>
          <type>sin</type>
        </parameters>
      </rftcomponent>
      <rftcomponent type="interpolator">
        <parameters>
          <outputBlockSize>4096</outputBlockSize>
          <filterCutoff>0.3</filterCutoff>
          <numTaps>16</numTaps>
        </parameters>
      </rftcomponent>
      <rftcomponent type="lowpassfirfilter">
        <parameters>
          <filterCutoff>0.3</filterCutoff>
          <numberTaps>16</numberTaps>
        </parameters>
      </rftcomponent>
    </parallel>

    <structure name="Filter">
      <parallel name="second" exclusive="false">
        <rftcomponent type="signalgenerator">
          <parameters>
            <blockSize>1024</blockSize>
            <samplingRate>44100</samplingRate>
            <frequency>440</frequency>
            <type>sin</type>
          </parameters>
        </rftcomponent>
        <rftcomponent type="lowpassfirfilter">
          <parameters>
            <filterCutoff>0.3</filterCutoff>
            <numberTaps>16</numberTaps>
          </parameters>
        </rftcomponent>
      </parallel>
    </structure>
  </reconfigurableradio>

```

Figure 5.10: XML description for two basic IRIS applications in parallel.

5.5.12 Embedded Signal-Processing Procedures

Multiple independent signal-processing applications can be described in a single XML configuration file. The XML configuration utility enables embedded reconfigurable radio/signal-processing structures to be specified in the configuration file. A more advanced level of IRIS implementation exists where a group of Radio Components may be encapsulated to appear as a single component in the XML configuration file. This is useful for reducing the complexity of the XML configuration allowing a much higher-level view of the reconfigurable radio system to be achieved. An illustrative example of this is the encapsulation of all of the Radio Components comprising the OFDM transmitter and OFDM receiver into two separate components as shown in Figure 5.11. The complex independent OFDM transmitter and receiver are encapsulated as two separate components shown in blue on this figure. Combined with an input and output Radio Components, this segment of the XML configuration may then be described using just four components. This would then enable very complex multi-transceiver implementations to be specified using a minimum number of component blocks in the XML configuration file.

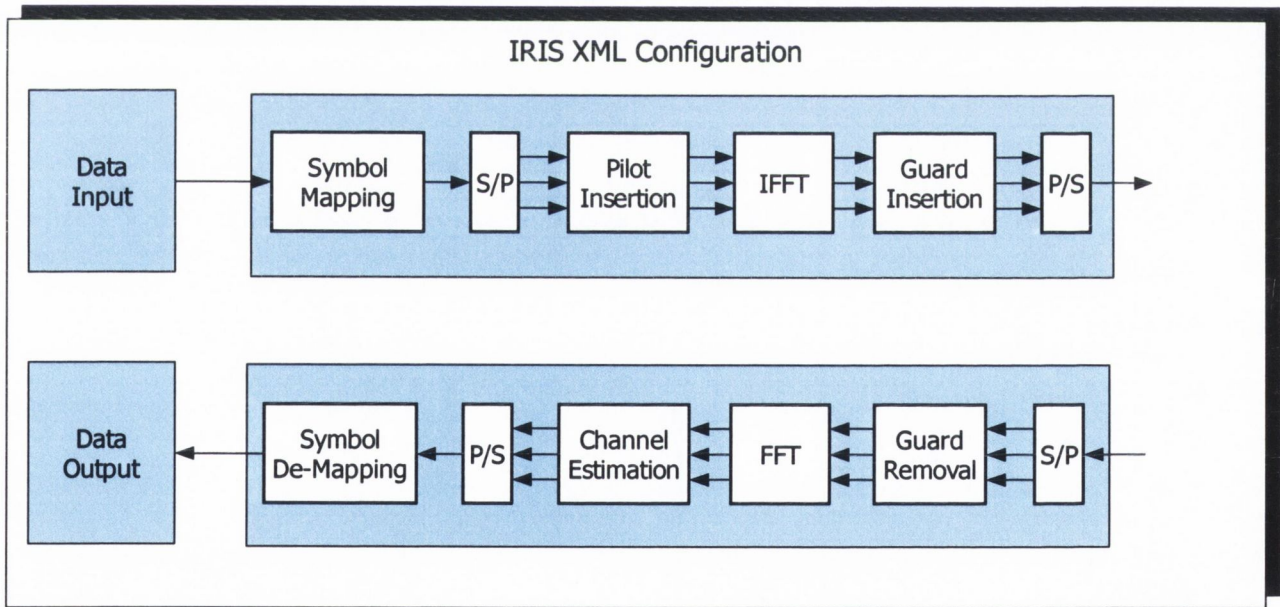


Figure 5.11: XML configuration example using embedded encapsulated components.

5.6 Hardware

The ADC/DAC and RF front-end chosen for the reconfigurable radio designs presented in this thesis are combined using a WaveRunner Plus 253 Peripheral Component Interconnect (PCI) transceiver board manufactured by Red River [RedRiver]. This board up-converts a baseband generated signal to an IF with a 3dB analogue bandwidth of 40 MHz centred at 70 MHz. Regarding the receiver, a channel with a maximum bandwidth of 8.6 MHz may be captured within this IF bandwidth and down-converted to a baseband signal. The baseband signal on both the transmission and reception signal paths may either in the form of I and Q, Amplitude or Phase information. The resolution of the ADC and DAC is 14-bits. This means that the 16-bit or 32-bit baseband transmission signal samples from the GPP are truncated before transmission. The dynamic range of the RF front-end is 90 dB, the sample clock rate is 93 MHz and the maximum power output is specified as -12dBm. The basic block diagram of the WaveRunner Plus 253 board is shown in Figure 5.12. A more comprehensive description of this board may be found by consulting the hardware reference manual [WaveRunner].

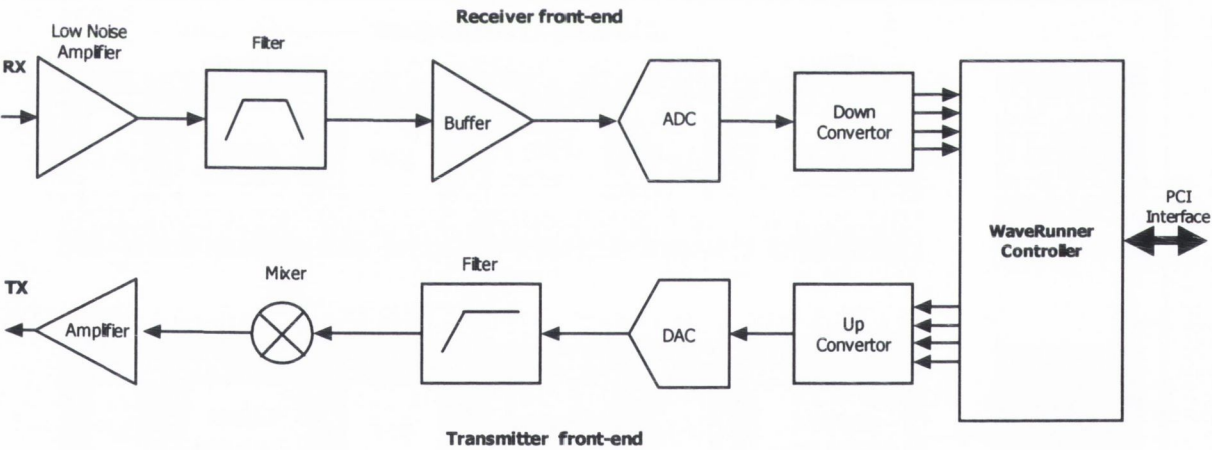


Figure 5.12: Block diagram of the WaveRunner Plus transmitter and receiver signal chains.

This combined ADC/DAC and up/down-conversion board may be configured by creating a Radio Component implementation of the WaveRunner Plus 253 transceiver. This means that it may be integrated into an IRIS XML radio configuration file and used for live transmission and signal capture. This board is capable of modulating and demodulating signals but in the context of this thesis, the reconfigurable radio carries out all of these tasks and this RF front-end is designed to be used only as baseband to passband and passband to baseband signal conversion unit. As this is an IF system, this signal may be

up-converted to other frequency bands including the 2.4 GHz ISM band using extra up/down-conversion RF hardware.

5.7 Conclusion

This chapter has provided an overview of the chosen reconfigurable radio development and implementation platform. This IRIS system has been designed specifically for the handling the control and management of reconfigurable radio applications. Now that the most important features of the IRIS system have been explained, the next stage is to describe how an actual OFDM reconfigurable radio is implemented using these principles and design methodology. The IRIS system enables the designer to compare a number of different variations of the same Radio Component in a radio application. This will prove to be a significant advantage when designing an enhanced OFDM reconfigurable transceiver.

6

OFDM IMPLEMENTATION

6.1 Introduction

The core objective of this chapter is to show how radio reconfiguration may be achieved using the IRIS and GPP platform. The fundamental basis for the design and creation of a reconfigurable radio is a software radio. Therefore the first half of this chapter describes how a software radio may be created using the IRIS framework. This chapter describes how this software radio may be developed into a reconfigurable radio. The term ‘reconfigurable OFDM transceiver’ is used in this chapter to denote the IF/baseband signal-processing functions of this radio system. These baseband signal-processing stages are the primary focus of this chapter. The importance of the IRIS component lifecycle descriptions and interface details described in the previous chapter will become apparent as this chapter progresses towards describing how a reconfigurable radio may be created.

Section 6.2 describes how a typical hardware radio may be translated to a software radio version, which forms the starting point for the reconfigurable radio version. A practical example of how a baseband OFDM modulator and demodulator may be created using the IRIS system is presented in Section 6.3. Section 6.4 then shows how this software radio evolves to form a reconfigurable radio. Section 6.5 summarises this chapter.

The results of the work presented in this chapter have been published in Proceedings of the Irish Signals and Systems Conference (ISSC) 2003 [Nolan2003c].

6.2 OFDM Software Radio

A standard software radio is the term used in this thesis to denote a software version of the IF/baseband signal-processing functions of a hardware radio. An example of this is

illustrated in Figure 6.1. The baseband functions of the OFDM transceiver, originally introduced in Chapter 2, are simply translated to software versions. These signal-chains of transceiver stages are shown in the red area in this figure to indicate that they are implemented purely in software. In this figure, all of the individual signal-processing stages for the OFDM modulator and demodulator are shown in blue. The OFDM modulator and demodulator signal chains are encapsulated in one large signal-processing block denoted by the white outlined rectangle.

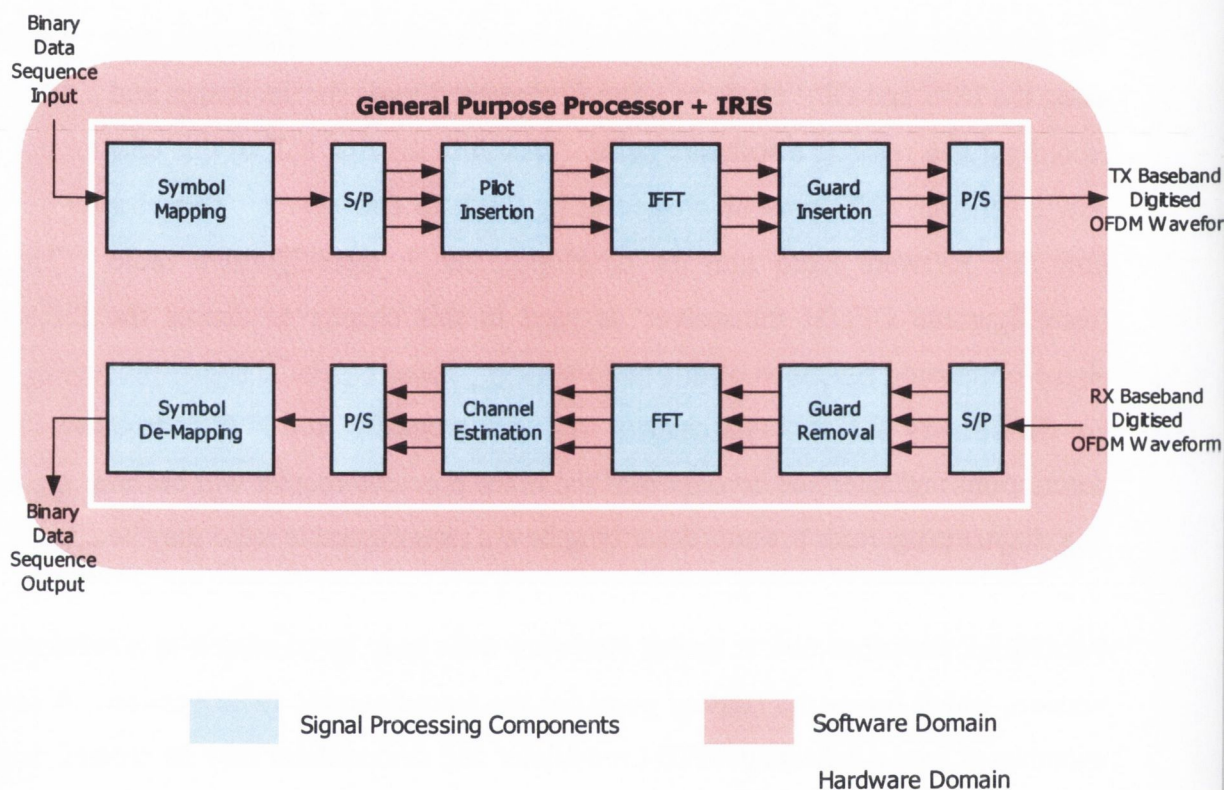


Figure 6.1: Software radio OFDM design example.

This software radio implementation is a translation of the IF/baseband processing functions of a hardware radio, but without any dynamic reconfiguration abilities. It is difficult to dynamically reconfigure a radio application that has been encapsulated as one large signal-processing entity. The reasons for this are that the parameters and internal functions may not be readily accessible or capable of being dynamically modified without having to re-compile and re-build the entire software programming code for this signal-processing stage. The objective therefore, is to separate this unwieldy encapsulated OFDM

transceiver implementation, shown in Figure 6.1, into its constituent parts, or sections. These individual sections may not be independent however. If one section of the reconfigurable transceiver is dynamically modified, the implications of this change may mean that one or more of the rest of the signal-processing sections may also need to be reconfigured. This may result in a cascading reconfiguration process throughout the entire transceiver design as each signal-processing stage in the radio design attempts to adapt to the changes that occurred in other areas of the radio.

For example, consider the case where the symbol-mapping stage, as shown in Figure 6.1, is reconfigured. By changing the modulation scheme, subsequent signal-processing stages in the radio may be affected by this change. Consider this example where the symbol-mapping stage, which was using BPSK modulation, is reconfigured to modulate information using QPSK. QPSK represents two bits per symbol whereas BPSK represents one bit per symbol. This means that for N input binary values, the modulator produces $N/2$ data symbols instead of N data symbols when BPSK was originally used. This means that the number of data symbols that must be converted to a number of smaller parallel sequences is now $N/2$. The number of these parallel sequences is determined by the number of sub-carriers, N_{sc} , that may be available for transmission. The Serial to Parallel (S/P) stage must therefore be reconfigured to create N_{sc} data subsets from the original data sequence. Therefore the length of each subset of data symbols must be halved. This is a simple example of how signal-processing components may have a mutual dependence. Reconfiguration and the issue of mutual dependence may be achieved using the IRIS Control Logic interface. An example of how Control Logic may be implemented will be presented in Section 6.4.

As described in the previous chapter, the IRIS framework enables functions and parameters within individual components to be accessed and dynamically modified using the Control Logic interface utilities. By adopting the IRIS approach for this reconfigurable OFDM transceiver, it is possible to dynamically reconfigure these individual signal-processing stages without having to re-compile each signal-processing stage. The IRIS framework, on which the software and reconfigurable radios are created, operates using a GPP. Microsoft Windows™ is the Operating System (OS) that will be used to support the IRIS framework.

6.2.1 IRIS Radio Component Creation

In order to implement a signal-processing algorithm or other type of function for use in the IRIS system, it is necessary to encapsulate the software programming code as a Radio Components for use with the IRIS framework. This section describes the processes that are required to create a Radio Component. As discussed in Chapter 5, Radio Components may be classed as either Input/Output (I/O), Digital Signal Processing (DSP) or Standalone components. Each of these component types has a generic programming structure or component 'skeleton', upon which the Radio Component is developed. The component creation procedures are almost exactly the same for all of other DSP components in the radio design. The completed reconfigurable OFDM transceiver will comprise several of these DSP components. To preserve the clarity of the explanation in this chapter, only the creation process for one component will be presented. This section therefore focuses on how a symbol-mapping component, which is classed as a DSP component, may be created.

6.2.1.1 Basic Building Blocks

Created using the C++ programming language, each Radio Component must be capable of being dynamically loaded and unloaded to/from the IRIS radio implementation. As the IRIS Framework was designed for use with the Windows™ OS, the Radio Components are created as dynamically interchangeable functional software objects called Dynamic Link Libraries (DLLs). These DLLs are the basic building blocks of the IRIS reconfigurable radio design. The Windows OS uses these DLLs as a means of sharing code and other information across multiple applications. An application such as IRIS may dynamically load this type of software object when the software functionality encapsulated within this object is required. Mackenzie's choice of using DLLs for the Radio Components also means that processor memory may be conserved since IRIS applications may share DLLs instead of having unique instances of the same software algorithm residing in memory at the same time. This means that the objectives of being able to create a new application, or modify and existing one may be met without having to stop the radio, re-compile and re-build a modified radio and then start a new transceiver application each time a new variation or additional functionality is required. The software design approach taken by Mackenzie is therefore a much more efficient method of implementing a reconfigurable radio. The IRIS framework manages the construction and co-ordination of the radio implementation leaving the designer to focus on the design and construction of the Radio

Components and Control Logic instead of the underlying processes that make a reconfigurable radio feasible.

6.2.1.2 Component Creation Procedure

The first task is to class the desired Radio Component as a DSP, I/O or a Standalone component. As discussed in the previous chapter, the generic class of functionality of each variation of Radio Component has a different level of abstraction from the core IRIS framework. For the symbol-mapping component example, the generic DSP Radio Component framework will be used.

6.2.1.3 Radio Component Functionality

The IRIS framework is capable of implementing many interfaces which are used for Radio Component and Radio Engine interaction, Control Logic and user-definable configuration files. A C++ header file which simply outlines the structure and supporting functions of the Radio Component according to the type of component is developed first.

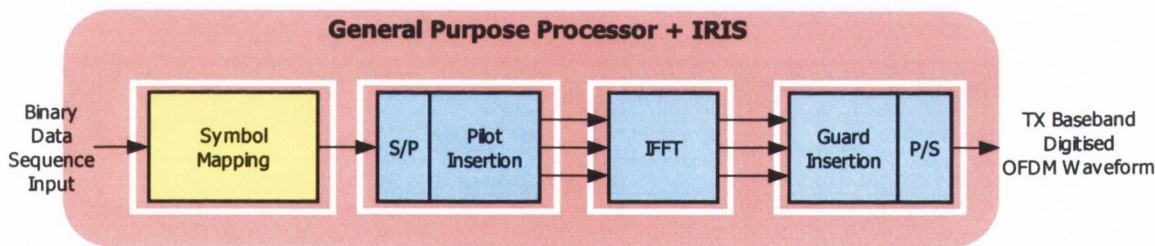


Figure 6.2: Separating the OFDM modulator into Radio Components.

The complete baseband signal-processing function of the OFDM modulator and demodulator as shown in Figure 6.1 must be separated into a number of individual Radio Components. Figure 6.2 focuses on the OFDM modulator section. This signal-chain has been separated into four groups of signal-processing stages and each of these groupings is encapsulated using a white outlined rectangle in this figure. Each of these four groupings will be implemented as a Radio Component. The Radio Component that will be developed in this section is the Symbol Mapping stage, which is highlighted in yellow in Figure 6.2.

It is important to note that the one or more of the signal-processing stages are grouped together to form one Radio Component as shown in Figure 6.2. For example, the serial to parallel (S/P) and pilot insertion stages will be implemented as a single Radio Component.

In terms of processing power and coding efficiency, it is sometimes more useful to combine signal-processing stages with low functionality and a high mutual dependence as a single Radio Component. The drawback of this approach is that the numbers of degrees of reconfiguration freedom can be reduced.

Figure 6.3 shows one example of how the symbol-mapping stage may be created as a Radio Component. This example uses the generic DSP component structure. The completed Radio Component will be a DLL and the internal operation and characteristics of this component may be altered using the exposed parameters and functions.


```

#include "DSPComponent.h"

//@component SymbolMapping component
//@author Keith Nolan
//@version 1.0
class SymbolMappingComponent : public DSPComponent
{
private:

    //      Start of exposed parameters //

    //@param number of bits per symbol
    //@default 2
    int numberOfBitsPerSymbol;

    //@param number of input bits
    //@default 1024
    int numberOfInputBits

    // End of exposed parameters //

    //      Other parameters used internally
    //      by the Radio Component.
    //      These are not exposed.

    int inputBlockSize;
    float *symbolarray;
    float *fdownsampledData;

public:

    SymbolMappingComponent();
    ~SymbolMappingComponent();
    virtual void GetDetails(ComponentDetails *details);
    virtual void CalculateOutputSignalFormat();
    virtual bool Init();
    virtual void Process(Signal input, Signal output);
    virtual void Destroy();

};

```

Figure 6.3: Symbol-mapping component function declaration example.

The parameters which are exposed by this particular Radio Component relate to the desired bit-to-symbol ratio and number of expected input binary values. Extra information is added to the class declaration of the Radio Component stating the default value of the parameters and whether they are required to be exposed by the completed DLL. This information is extracted from the class description using a scripting engine that is part of the IRIS framework. The prefix “//@” denotes the information that will be parsed by this scripting engine. This IRIS scripting engine uses this information create extra files that are

both intuitive to the designer and easily understood and managed by the IRIS Component Manager.

Consider this declaration as shown in Figure 6.3:

```
//@param number of bits per symbol
//@default 2
int numberOfBitsPerSymbol;
```

This is a declaration of the parameter used to specify the number of bits per symbol that a modulated data symbol uses. This particular parameter has been chosen to be a 32-bit integer type in this case.

The declaration:

```
//@param number of bits per symbol
```

denotes that this parameter will be exposed by the DLL and may be altered using the external XML configuration file when the component DLL is completed.

The declaration:

```
//@default 2
```

is used to denote the default value of this parameter, which is 2 bits per symbol in this case.

In a similar fashion, the second exposed parameter in this component header file, which relates to the number of input bits, is described by:

```
//@param number of input bits
//@default 1024
int numberOfInputBits
```

The number of bits will be modulated by this component is also specified using a 32-bit integer in this case. The default value of this parameter is 1024. This means that this symbol-mapping component expects an input signal of size 1024 bits (128 bytes). It is generally unnecessary to specify an input-size parameter for all of the Radio Components in the signal chain. The reason for this is that the IRIS Radio Engine informs each of the

Radio Components that follow this initial symbol-mapping stage of the expected number of input values for each subsequent component.

The constructor and destructor function declarations also highlighted in Figure 6.3 are not only common to all of the Radio Component class declarations but in fact are integral parts of all C++ OOD class declarations. In this example, the constructor function declaration has the same name as the class. In the example presented in Figure 6.3, the class name is ‘SymbolMappingComponent’. The constructor is therefore denoted as:

```
SymbolMappingComponent();
```

This constructor is the first function invoked automatically by the application when an instance of a class is instantiated and may be used to allocate memory and initialise parameters used in the particular design. The destructor is generally the final function that is also automatically invoked by the application when the instance of the class is terminated or destroyed and may be used to release memory and other resources in addition to any other miscellaneous ‘clean-up’ tasks specified by the designer. In this example, the destructor is:

```
~SymbolMappingComponent();
```

The functions relating to the life-cycle of the Radio Component are specified in the class declaration also. There are other functions that have already been created by the IRIS scripting engine (in other words, the functions that are not user-defined). These are also required in order to create the final component DLL and are contained in a separate header file relating to the Radio Component type. As this particular component is a DSP component, the header file that contains these extra function declarations is included using the declaration:

```
#include "DSPComponent.h"
```

The following functions:

```
virtual void GetDetails(ComponentDetails *details);  
virtual void CalculateOutputSignalFormat();
```

are used to retrieve the details of the Radio Component instance and set the correct data-type format of the information processed by the component, respectively. The bodies of these functions are generated by the IRIS scripting engine automatically.

The most important functions from a radio designer's point of view are:

```
virtual bool Init();  
virtual void Process(Signal input, Signal output);  
virtual void Destroy();
```

As described earlier, the `Init()` function encapsulates the initialisation procedures for each of the components. This may be used to initialise the arrays and miscellaneous parameters used as part of the radio design. For this symbol-mapping component example, the `Init()` is used to create and initialise the component class instance. This is the first designer-specific function that is invoked after the constructor of a class instance completes. The `Init()` function is automatically invoked by the IRIS Radio Engine each time a new instance of a Radio Component class is created.

The `Process()` function is a designer-specified function which is used to carry out the core objectives of the component. In this symbol-mapping example, this actual symbol mapping algorithm is encapsulated in this function. As discussed in the previous chapter, one of two variations of the `Process()` function may be used in a Radio Component implementation depending on whether the size of the processed data array is the same as the input array or if the processed data array is larger than the original input array. For the former, the *in-place* variant may be used, which means memory for the output array does not have to be allocated at the expense of overwriting the original input data. The *not-in-place* variant may be used when the size of the output array is expected to differ in size than the original input array and/or when preservation of the original input data array is required. This symbol-mapping component example uses the *not-in-place* version of the `Process()` function as both input and output memory pointers are specified as input parameters.

An example of how the `Process()` function may be implemented for the symbol-mapping Radio Component is shown in Figure 6.3.

```
void SymbolMappingComponent::Process(Signal input,Signal output)
{
    //Pointer to array of input binary values
    char *input = (char*)input.data;

    //Pointer to array of output complex-valued data symbols
    SCplx *output = (SCplx*)input.data;

    //    Modulate the input binary values

    //    The number of modulated data symbols is dependent
    //    the size of
    //    the input array and numberOfBitsPerSymbol

    output = Modulate(input,numberOfBitsPerSymbol);
}
```

Figure 6.4: `Process()` function used for the Symbol Mapping Radio Component example.

This function has two input parameters. These are pointers to the memory location of the array of input binary values, which will be modulated, and the pointer to the memory location of the destination array. The destination array will be used to store the complex-valued data symbols. The function `Modulate(input,numberOfBitsPerSymbol)` is invoked to actually perform the symbol-mapping (modulation). This function is defined by the radio designer and in this example, two input parameters are required. The pointer to the memory location of the input binary values is required and the exposed function relating to the number of bits per symbol (in effect, this determines the modulation scheme) are both passed to this function as input parameters. The end results of this function, which are the modulated data symbols are then written to the memory location specified by the pointer `output`. Following the completion of this `Process()` function, the IRIS Radio Engine manages the transferral of processed information to the next Radio Component in the signal chain.

The final function in the life-cycle interface as shown in Figure 6.3 is the `Destroy()` function. This is another user-definable function that may be invoked manually in order to release internal and external resources when the radio implementation or Radio Component is being shutdown. This is also invoked to safely remove a component from an operating radio implementation.

6.3 Example: OFDM Modulator and Demodulator

6.3.1 Complete OFDM implementation using IRIS

The OFDM modulator and demodulator signal chains as originally introduced in Chapter 2 and presented in this chapter as Figure 6.1 are further illustrated in Figure 6.5. The white outlined rectangles in this figure show how the signal-processing stages have been encapsulated as DSP Radio Components. Each of these Radio Components is implemented using similar procedures as for the symbol-mapping component example. The differences however, lie with the number and type of parameters and the signal-processing functions that may be invoked when the core `Process()` function of each Radio Component is invoked.

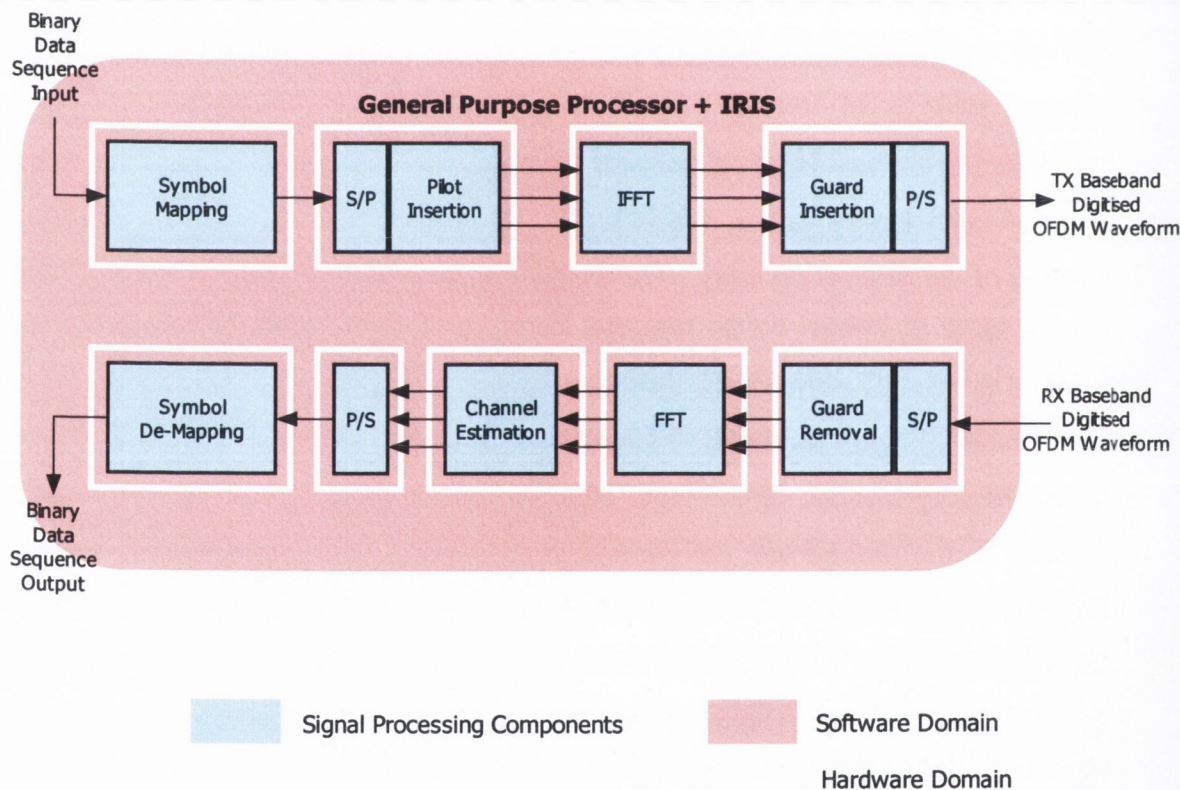


Figure 6.5: Complete OFDM modulator and demodulator comprised of Radio Components.

Instead of presenting the programming code for all of these Radio Components, it is more useful to show some of the typical parameters, ports and events that may be exposed by these components. This will then lead onto a description of how these Radio Components may be dynamically reconfigured using these accessible component entities.

6.3.1.1 Exposed Parameters, Ports and Events

A listing of the exposed parameters, ports and events associated with each Radio Component is very useful for the management of all of the components. It is perhaps more intuitive than providing programming code listings in order to explain the accessible parts of each component. The exposed parameters, ports and events for three of the Radio Components comprising part of the OFDM modulator and demodulator implementation as shown in Figure 6.5, will be presented.

Beginning with the symbol-mapping Radio Component described in the previous section, Table 6.1 lists the exposed parameters, ports and events associated with this component.

Table 6.1: Table of exposed parameters, ports and events for the symbol-mapping component.

Symbol Mapping	Description
Parameters numberOfBitsPerSymbol numberOfInputBits	The number of bits represented by one data symbol
Ports n/a	
Events n/a	

As previously discussed, this symbol-mapping Radio Component implementation only has two exposed parameters and no ports or events.

The next Radio Component being examined is the IFFT stage of the OFDM modulator. The exposed parameters, ports and events associated with this component are listed in Table 6.2.

Table 6.2: Table of exposed parameters, ports and events for the IFFT Radio Component.

IFFT	Description
Parameters	
IFFTsize	The number FFT bins used.
Ports	
n/a	
Events	
changedIFFTsize	Notifies other Radio Component/Control Logic that the FFT size has been changed.

The most important parameter in this component states the number of FFT bins that will be used to transform the input data. This parameter is denoted as `IFFTsize`. In this example, the event `changedIFFTsize` is specified. This event may be triggered when other Radio Components request a change of FFT size. An action like this may occur if a low bandwidth OFDM application is required with only a few sub-carriers being employed or if a scheme such as DVB-T is implemented. For this latter case, the FFT size would have to be increased significantly.

The third Radio Component being examined is the FFT stage of the OFDM demodulator. In this case, the main exposed parameter is the FFT size denoted by `FFTsize`. One port is specified for this component. As well as processing the main stream of data arriving from the previous component in the signal chain, other Radio Components may send data to the FFT stage via this port denoted as `secondaryInput`. This may occur if the power spectrum of a signal is required. This FFT stage is a fundamental part of this operation and therefore could facilitate this task in addition to processing the main input signal sequences. The event `processedPortData` is triggered when the data that arrived via the port has been processed and the Radio Component/Control Logic function that requested this operation must be notified of its completion.

Table 6.3: Table of exposed parameters, ports and events for the FFT stage in the OFDM demodulator.

FFT	Description
Parameters FFTsize	The number FFT bins used.
Ports secondaryInput	This allows separate data sequences to be processed
Events processedPortData	Notifies other Radio Component/Control Logic that the port data processing has completed

The IRIS XML configuration description enables the values of the exposed parameters to be designated before the radio implementation commences operation in addition to describing the initial radio configuration. Many other parameters and variables are required as part of the component programming code but these may be ‘hidden’ inside the Radio Component object. In addition to the exposed parameters that may be configured by the user, the radio designer may have to deal with any ports and events that may be part of a Radio Component. As shown in Table 6.3, the FFT component in this example specifies one port and one event. Information sent to this port must be handled using a user-specified port handling routine.

6.3.1.2 Port Handling Routine

The FFT component in the example shown in Figure 6.5, may also process information that is received through a port. A user-defined function is required to react to data that is passed to the port and then send this data as an input to the `Process()` function. The function that manages this operation is called a port handling routine and may be declared as:

```
ProcessPortData(int portID, unsigned char* data, int length)
```

The input parameters are the identity of the port where the input data arrived from, the input data itself and the length of the input data array, respectively.

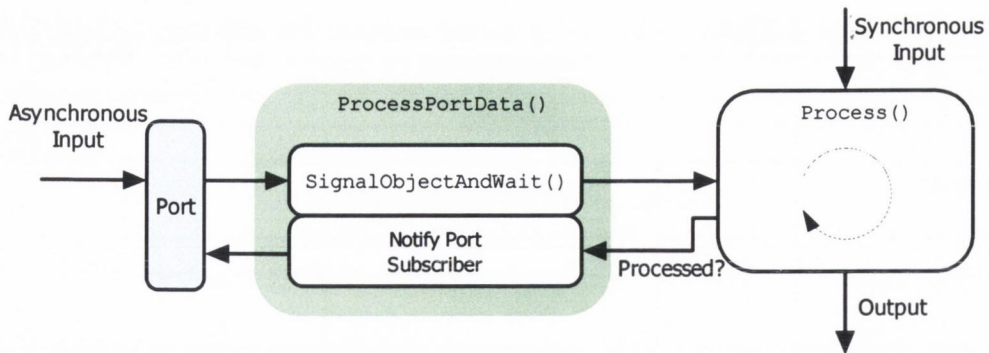


Figure 6.6: Asynchronous and synchronous input-processing concept.

```

bool FFTComponent::ProcessPortData(int portID, unsigned char* data, int length)
{
    if(portID == PORT_PROCESSDATA)
    {
        //Allocate memory and copy the received data
        dataToProcess = new unsigned char(length);

        //Set the length of the received data
        dataToProcessLength = length;
        memcpy(dataToProcess,data,length);

        //Signal that data is ready to be processed
        //and wait until this operation has completed
        signalObjectAndWait(hEventWait,hEventComplete,INFINITE,FALSE);

        //Release allocated memory
        delete [] dataToProcess;
        dataToProcess = NULL;
        dataToTransmitLength = 0;

        //Trigger event to notify the external port subscriber(s)
        //that the data has been processed
        ActivateEvent(EVENT_DATAMODULATED,length);

        return TRUE;
    }
    return FALSE;
}

```

Figure 6.7: Port handling routine example for the FFT Radio Component.


```

<reconfigurableradio>

  <description>
    <name>OFDM modulator</name>
    <author>Keith Nolan</author>
    <comment>Implements an OFDM Modulator</comment>
  </description>

  <structure name="OFDM modulator">
    <rfcomponent type="binarySignalGenerator">
      <parameters>
        <blockSize>1024</blockSize>
        <type>binary</type>
      </parameters>
    </rfcomponent>
    <rfcomponent type="symbolMapper">
      <parameters>
        <numberOfBitsPerSymbol>2</numberOfBitsPerSymbol>
        <numberOfInputBits>1024</numberOfInputBits>
      </parameters>
    </rfcomponent>
    <rfcomponent type="serialToParallelAndPilotInsertion">
      <parameters>
        <numberOfSubcarriers>64</numberOfSubcarriers>
        <numberOfPilotCarriers>4</numberOfPilotCarriers>
      </parameters>
    </rfcomponent>
    <rfcomponent type="IFFT">
      <parameters>
        <IFFTsize>128</IFFTsize>
      </parameters>
    </rfcomponent>
    <rfcomponent type="GuardInsertionAndParallelToSerial">
      <parameters>
        <guardLength>16</guardLength>
      </parameters>
    </rfcomponent>
    <rfcomponent type="output">
      <parameters>
        <scalingFactor>1</scalingFactor>
      </parameters>
    </rfcomponent>
  </structure>
</reconfigurableradio>

```

Figure 6.8: XML configuration file for a signal generator and an OFDM modulator.

Figure 6.6 is an illustration of the concept of a port and the associated port handling routine. Asynchronous input to the Radio Component is achieved using the port interface. The information is sent to the port by a designer-specified Control Logic function. The Control Logic in this case may be originally obtained the information from another Radio Component in the signal chain and thus facilitates the transfer of information between Radio Components. The function `ProcessPortData` copies this data to memory and

passes it to the main `Process()` function in the Radio Component. Upon completion, the port subscriber (i.e. the Control Logic function connected to the port) is notified using an event. An example of the C++ programming function that implements this `ProcessPortData` procedure is shown in Figure 6.7.

6.3.2 Completing the OFDM modulator Implementation

Each of the Radio Components encapsulated in Figure 6.5 may be structured using a XML configuration file. An example of the XML file configuration for the OFDM modulator part of the radio implementation shown in Figure 6.2 is shown in Figure 6.8. The exposed parameters associated with each Radio Component in the design may be specified using this file. The signal flow in this case is from the top-most specified Radio Component to the component at the end of the XML configuration file. This radio description also includes a binary signal generator component. This provides the data for the OFDM modulator in this example. An output device is also specified at the end of the XML configuration file and this may be designed to pass the generated OFDM information to an ADC and the RF front-end.

The OFDM demodulator is specified in a similar manner and the both the OFDM modulator and demodulator are implemented as parallel structures operating asynchronously. This is achieved by creating two asynchronous radio structures in the XML configuration file. A truncated example of this is shown in Figure 6.9. In this figure, one reconfigurable radio structure implements the OFDM modulator signal chain that was presented in Figure 6.2 and in XML configuration file form in Figure 6.8. A second reconfigurable radio structure describes the OFDM demodulator. The Radio Components that may be used to implement this demodulator are illustrated in Figure 6.5.


```

<reconfigurableradio>

  <description>
    <name>OFDM modulator and demodulator</name>
    <author>Keith Nolan</author>
    <comment> OFDM Modulator and Demodulator</comment>
  </description>

  <structure name="OFDM modulator">
    <rfcomponent type="binarySignalGenerator">
      <parameters>
        <blockSize>1024</blockSize>
        <type>binary</type>
      </parameters>
    </rfcomponent>

    ....truncated

  </structure>

  <structure name="OFDM Demodulator">
    <rfcomponent type="input">
      <parameters>
        <blockSize>1024</blockSize>
      </parameters>
    </rfcomponent>
    <rfcomponent type="serialToParallelAndGuardRemoval">
      <parameters>
        <numberOfSequences>64</numberOfSequences>
        <guardLength>16</guardLength>
      </parameters>
    </rfcomponent>

    ....truncated

  </structure>

</reconfigurableradio>

```

Figure 6.9: Truncated IRIS XML OFDM modulator and demodulator configuration file example.

The IRIS system includes a Graphical User Interface (GUI) that may be used to edit the configuration of the radio, edit the values of the exposed parameters of each Radio Component in the radio implementation and create new IRIS radio applications. This removes the need for the radio designer/user to have to modify the source code used to create these components. As a result, rapid manual modifications may be made to the radio design allowing fast prototyping and testing.

6.4 Enabling Reconfiguration

6.4.1 Overview

Control Logic is essentially external programming code that is capable of altering the radio configuration, where the initial radio configuration has been described in XML form. The Control Logic capabilities of IRIS are used to reconfigure, restructure and control the reconfigurable radio. As discussed previously in this chapter, the Control Logic interface used in the IRIS framework enables the exposed parameters of each of the Radio Components in a particular IRIS radio design to be accessed and dynamically modified. This interface enables the components to remain independent of the specific IRIS application yet facilitate the mutual-dependence that may exist between components as previously discussed. The Control Logic programming code is application-specific and is not mandatory for every IRIS radio implementation unless dynamic reconfiguration and/or dynamic control of one or more Radio Component is required.

6.4.2 Dynamic Parameters

The C++ header file relating to the symbol-mapping Radio Component is shown again in Figure 6.10. The difference between this version of the component header file and the original version as previously described in Figure 6.3 is that an extra piece of script code has been added to the parameter declarations. A dynamically modifiable parameter may be denoted using the prefix “`///@dynamic”`. In this example, the two exposed parameters, which may be now dynamically reconfigured, are `numberOfBitsPerSymbol` and `numberOfInputBits`.

When the Radio Component is being initialised, the IRIS parser and scripting engine automatically attaches extra IRIS-specific functions to each of the dynamic parameters. These functions update the parameter when the Control Logic initiates a modification and also notify the Radio Engine that a parameter has been modified. The Radio Engine must be notified because a parameter modification may mean that one or more of the other dependent parameters within the Radio Component will have to be updated.


```

#include "DSPComponent.h"

//@component SymbolMapping component
//@author Keith Nolan
//@version 1.0
class SymbolMappingComponent : public DSPComponent
{
private:

    //      Start of exposed parameters //

    //@param number of bits per symbol
    //@default 2
    //@dynamic
    int numberOfBitsPerSymbol;

    //@param number of input bits
    //@default 1024
    //@dynamic
    int numberOfInputBits

    // End of exposed parameters //

    //      Other parameters used internally
    //      by the Radio Component.
    //      These are not exposed.

    int inputBlockSize;
    float *symbolarray;
    float *fdownsampledData;

public:

    SymbolMappingComponent();
    ~SymbolMappingComponent();
    virtual void GetDetails(ComponentDetails *details);
    virtual void CalculateOutputSignalFormat();
    virtual bool Init();
    virtual void Process(Signal input, Signal output);
    virtual void Destroy();

};

```

Figure 6.10: Enabling parameter reconfiguration using the '@dynamic' tag.

6.4.3 Reconfiguration Using Control Logic

The external Control Logic programming code may be used to modify one or all of the dynamic parameters. The Control Logic application must first locate the Radio Components and exposed parameters of interest. Figure 6.11 illustrates the principle of how IRIS Control Logic programming may be used to access and modify the exposed parameters in each of the Radio Components and to enable information to pass to/from

Radio Components. Designer-specified programming routines within the Control Logic entity may be used to compute updated parameter values based on the values of the static and dynamic parameters in one or more of the Radio Components. Each of the affected Radio Components in the design may then be updated. The Control Logic interface functions also enable Radio Components to be removed and replaced during the operation of the radio implementation.

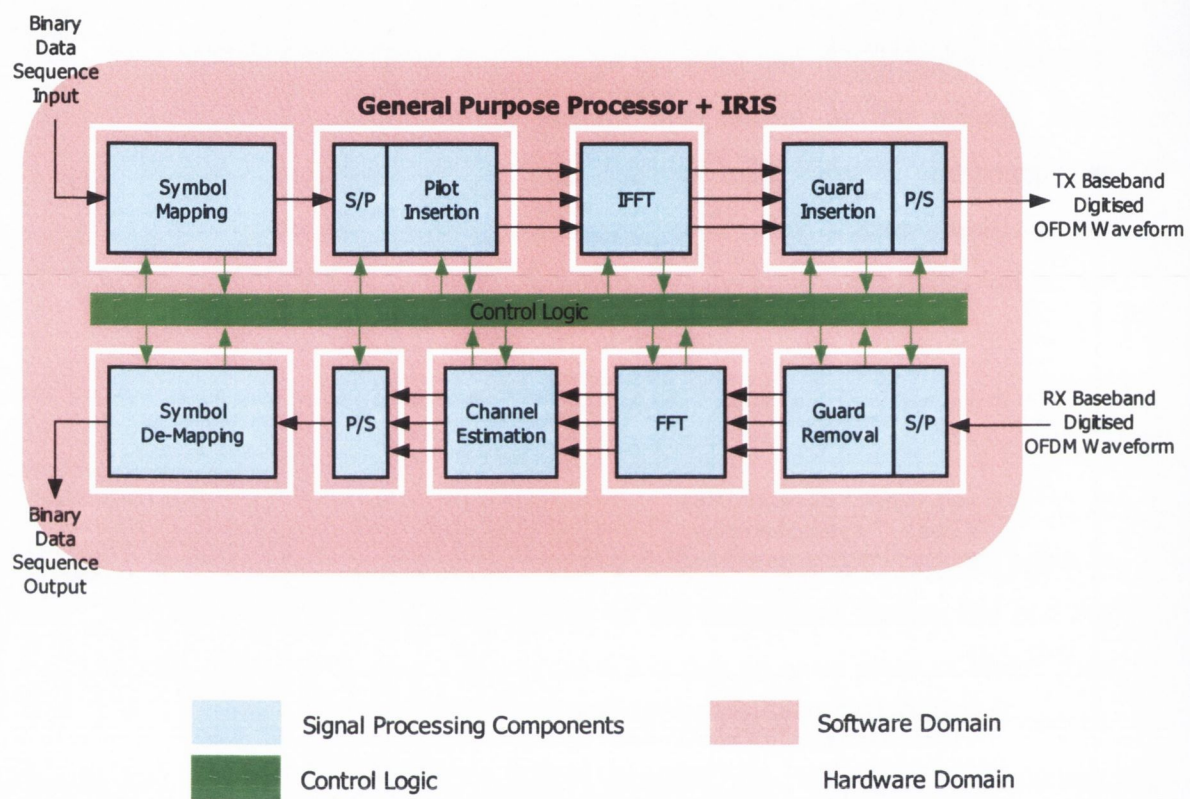


Figure 6.11: Reconfiguration using Control Logic.

A truncated version of a sample Control Logic programming code class and function declaration file for the OFDM modulator example is shown in Figure 6.12. This defines a Control Logic application controller that enables modification of the IRIS radio configuration if required.


```

//Invoked to initialise Control Logic
virtual bool AttachToComponents()
{
    HANDLE_COMPONENT hComponent1;
    HANDLE_COMPONENT hComponent2;
    HANDLE_PARAMETER hParameter1;
    HANDLE_PARAMETER hParameter2;
    HANDLE_PARAMETER hParameter3;

    //Find the SymbolMapper component
    hComponent1 = engine->FindComponent("SymbolMapper");

    //Find the numberOfBitsPerSymbol parameter
    hParameter1 = engine->FindParameter("numberOfBitsPerSymbol");

    //Find the numberOfInputBits parameter
    hParameter2 = engine->FindParameter("numberOfInputBits");

    //Find the IFFT component
    hComponent = engine->FindComponent("IFFT");

    //Find the IFFTsize parameter
    hParameter3 = engine->FindParameter("IFFTsize");

    //Attach Control Logic programming routines to the events //specified in
    the component
    //C++ header file
    engine->SubscribeToComponentEvent("IFFT", "changedIFFTsize", (int)this,
    Routine1);
}

```

Figure 6.12: Excerpt from the Control Logic application for two Radio Components.

The Control Logic application must first ‘attach’ itself to the Radio Components of interest and then locate the exposed parameters that were labelled as being ‘dynamic’, as previously discussed. In this example shown in Figure 6.12, Control Logic is used to locate the symbol-mapping and IFFT Radio Components used in the OFDM modulator example.

The line:

```
engine->FindComponent("SymbolMapper");
```

attempts to locate the symbol-mapping component. If this function succeeds, the handle `hComponent1` is used to reference this component.

The next line of the application is:

```
engine->FindParameter("numberOfBitsPerSymbol");
```

This command attempts to locate the exposed dynamic parameter, `numberOfBitsPerSymbol` in the symbol-mapping component. This procedure is repeated for all of the Radio Components that will be reconfigured using the Control Logic application.

As stated in Table 6.2, the IFFT component uses an event that may be triggered when some set of criteria within the IFFT component is satisfied. The line:

```
engine->SubscribeToComponentEvent("IFFT", "changedIFFTsize"  
(int)this, Routine1);
```

is required to ensure that the Control Logic will be capable of listening and responding to the event `changedIFFTsize`. A function which is invoked when this event is triggered is also specified as an input parameter. In this case, the function is called `Routine1` and the tasks that this function may carry out are designer-specified.

6.5 Summary

This chapter has discussed the principles of how a radio may be created and subsequently dynamically reconfigured using the IRIS framework Control Logic interface functions. All of the dynamic parameters exposed in a Radio Component may potentially be accessed and modified using Control Logic programming code. This programming code may invoke user-defined routines and even extra signal-processing tasks outside of the main IRIS reconfigurable radio structure. By enabling radio reconfiguration in a straightforward and uncomplicated fashion, as is the case with the IRIS framework, the radio designer is free to quickly develop and test new communication techniques and concepts with the aim of improving both the performance and capabilities of the radio. This chapter concludes the second main part of this thesis.

7

FRAME SYNCHRONISATION

7.1 Introduction

This chapter focuses on frame synchronisation in an OFDM system. Frame synchronisation is a means of estimating where the start of an OFDM frame occurs in a received OFDM signal. In an OFDM transceiver system, it is imperative that the receiver is capable of estimating where the start of an OFDM frame occurs in a signal sequence in order to correctly extract the sequence of OFDM symbols comprising the modulated information from the source transceiver. Frame synchronisation is important as misalignment between the receiver and transmitter may result in ISI and Inter-Carrier Interference (ICI) due to the possible loss of orthogonality between the sub-carriers. Section 7.2 places the work in the context of this thesis. In Section 7.3 a review of some of the main algorithm proposals that have lead to feasible frame synchronisation on software radio/reconfigurable radio platforms is described. In Section 7.4 a solution is identified. Sections 7.5 and 7.6 are concerned with the implementation and evaluation of this solutions and Section 7.7 concludes.

The results of the work presented in this chapter have been published in Proceedings of the IEEE Global Telecommunications Conference (IEEE Globecom) 2003 [Nolan2003a], and in Proceedings of the International Association of Science and Technology for Development (IASTED) Signal Processing, Pattern recognition and Applications (SPPRA) 2003 technical conference [Nolan2003d].

7.2 Placing the Work in Context

Figure 7.1 illustrates the main signal-processing functions required for an implementation of an OFDM transceiver that was originally presented in Chapter 2. It was shown in this chapter and by Eq. [2.8], that an OFDM frame is created by converting M parallel

sequences of cyclically-extended OFDM symbols into a serial sequence of M OFDM symbols. Each OFDM frame is separated in the time domain from subsequent OFDM frames by a frame guard interval specified by the radio designer and has a duration greater than the expected channel impulse response. The Serial to Parallel (S/P) conversion stage of the OFDM demodulator in this system must therefore partition the received sequence of M OFDM symbols contained in a frame back into M OFDM symbol sequences in parallel. Each of the OFDM symbols must be aligned with the start of each FFT array in the time domain in order to ensure that the information contained in each OFDM symbol may be de-multiplexed correctly. In order to achieve this therefore, the S/P conversion stage must begin partitioning the incoming OFDM signal sequence at the correct point of the incoming signal sequence. The correct starting point is the start of each OFDM frame. Therefore, OFDM frame synchronisation is required to locate the start of the OFDM frames. This stage is a crucial part of the OFDM demodulation process.

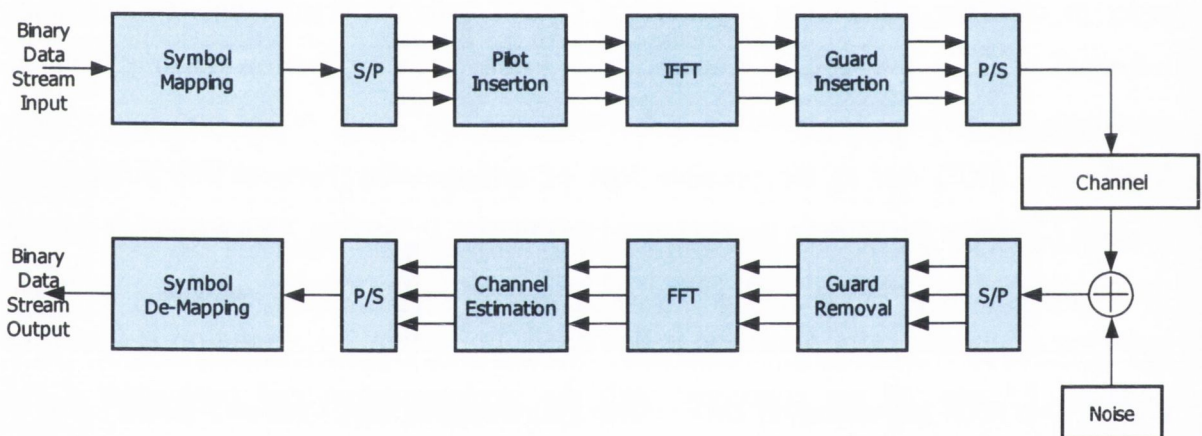


Figure 7.1: Block diagram of a basic OFDM transceiver.

Figure 7.2 illustrates the relevant parts of a modified version of the OFDM transceiver based on the basic transceiver block diagram shown in Figure 7.1. Regarding Figure 7.2, the signal-processing stages in yellow comprise the main areas of interest in this section. This model has been updated to include a means of establishing frame synchronisation involving the insertion of a frame synchronisation pilot system at a specific time before the start of a new OFDM frame. An OFDM frame is formed during the conversion of M OFDM symbols in parallel to a serial OFDM symbol sequence of M OFDM symbols. The receiver has been developed further to include a frame synchronisation stage that searches the incoming signal sequence for the frame synchronisation pilot symbol denoting the start of an OFDM frame.

It will be shown that frame synchronisation and frequency synchronisation are not necessarily independent tasks. As a result, the objective of low-complexity carrier-frequency offset estimation is also incorporated into the optimum frame-synchronisation algorithm. The main objective is to obtain a robust and reliable frame synchronisation algorithm resulting in a minimum transmitted signal sequence overhead. Additionally, as signal demodulation using DSP techniques has advanced, another important objective is to choose a frame synchronisation technique that is low in complexity and will not result in a high signal-processing load on the target device.

Frame synchronisation is used for all of the OFDM-based transmission schemes described in Chapter 2. For continuous transmission schemes such as DAB and DVB-T, the frame duration and frame periodicity is generally constant thus maintaining frame synchronisation is not a complex task following initial acquisition. However, frame synchronisation for OFDM transmissions that may occur irregularly or in bursts (burst-mode transmission) require a robust and rapid frame synchronisation technique. This scheme must enable the index of the frame start in the intercepted signal sequence to be estimated and the OFDM symbols contained in the frame to be demodulated before the next frame is intercepted.

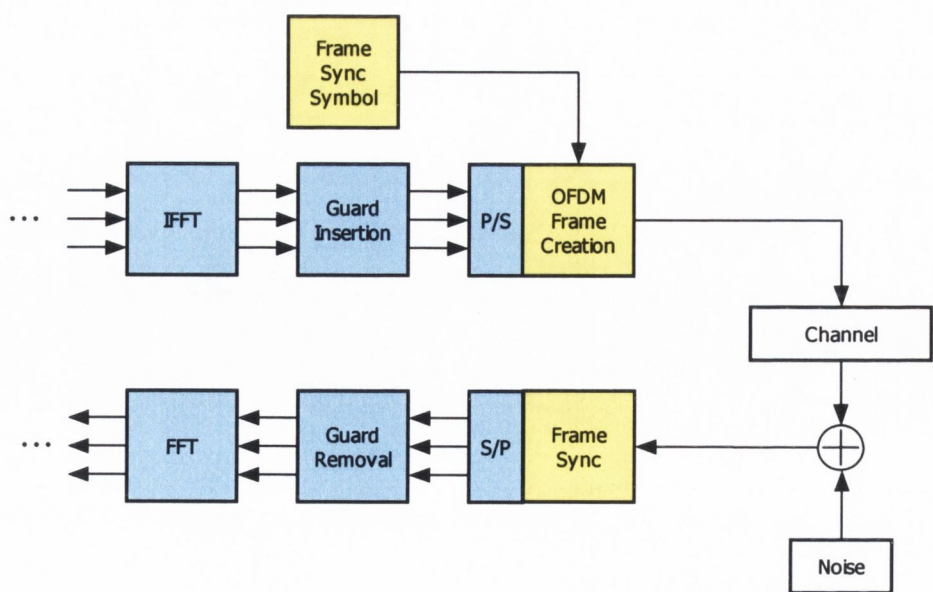


Figure 7.2: Modified OFDM transceiver including frame synchronisation stages.

The directly relevant research work associated with this will now be presented, leading to a description of the chosen technique that forms a basis for the frame and frequency synchronisation technique used for the reconfigurable transceiver system presented in this thesis.

7.3 Review of Frame Synchronisation Techniques

Several research papers in the last few years have focused on hardware implementations of OFDM systems and synchronization issues but the research findings may be adapted to use in a reconfigurable radio implementation. This section presents a brief overview of some of the main contributions to frame (and carrier-frequency) synchronisation.

Chevillat, Maiwald *et al* [Chevillat1987] proposed a rapid equalizer training scheme which compensates for amplitude and phase distortion introduced by the communications channel and does not necessitate a timing preamble. This scheme is of importance because it deals specifically with Constant Amplitude Zero Auto-Correlation (CAZAC) sequences such as OFDM. CAZAC sequences are complex valued sequences with constant amplitude values and the narrow regions of concentrated power spectral density values, or spectral lines, have equal amplitude. Of course, the effectiveness of this scheme relies on the linear properties of the transmitter and receiver amplifiers, especially as the transmission bandwidth increases.

Choi [Choi1990] examined frame alignment in digital carrier systems using the Synchronous Optical Network (SONET) Synchronous Transport Signal-3 (STS-3) fibre optic transmission standard as an example. Although this work does not target OFDM systems specifically, this paper does present an analysis of the issues which arise further up in the bit-processing layers of the signal-processing chain when incorrect data frame synchronisation occurs. The data frame synchronisation method chosen was to embed known binary sequences into overall data sequence and use them to mark the start of data frames, when processed by the destination device. This process is analogous to many of the frame synchronisation techniques used in OFDM where known pilot symbols may be transmitted before the start of a frame in order to enable receiver synchronisation.

One of the first significant research findings regarding synchronisation for OFDM systems, which is of direct relevance to the research work presented in this thesis was published in

1993. One of the significant factors of this work is that Warner and Leung [Warner1993] proposed a correlation-based frame synchronisation scheme designed for mobile fading wireless channels. The two main criteria for Warner’s proposed frame synchronisation scheme are that frame synchronisation is achieved on a per-frame basis, and due to spectrum limitations, the transmission overhead had to be minimal. This scheme is based on correlating known pilot symbols with the corresponding versions of the received pilot symbols and then denoting the index of the peak correlator output value as the position of the start of the frame. It was found that synchronisation using correlation-based techniques is dependent on the particular sub-carriers used for pilot symbol transmissions due to the sub-carrier side-lobe characteristics. The reported BER performance degradation of this scheme over an ideal synchronisation case was less than 1.5 dB employing a bandwidth overhead of less than 10%.

In 1995, Sandell, van de Beek and Börjesson [Sandell1995] presented their findings regarding a Maximum Likelihood (ML) OFDM timing and frequency synchronisation without using pilot symbols. Their proposal involved exploiting the cyclic prefix as a means of establishing frame synchronisation and carrier frequency offset estimation. In this paper, Sandell’s definition of an OFDM frame comprises one cyclically-extended OFDM symbol, whereas in this thesis, an OFDM frame is defined as a series of a number of cyclically-extended OFDM symbols. A frame guard is then appended (or pre-pended) to this OFDM frame. The term ‘OFDM symbol’ as understood in the context of this thesis is understood by Sandell as an OFDM ‘frame’. As explained in Chapter 2, a cyclic prefix may be obtained by pre-pending the last N_{GI} samples of an OFDM symbol to the same OFDM symbol, where N_{GI} denotes the number of guard-interval signal samples. The ML OFDM symbol synchronisation scheme proposal relies on correlating the N_{GI} cyclic prefix samples with the last N_{GI} samples of the OFDM symbol. The performance of this algorithm was considered for AWGN channels only. This particular scheme would not be suitable for the reconfigurable radio implementation described in this thesis due to the per-OFDM symbol synchronisation requirements and resulting processing power required to implement this.

Van de Beek, Sandell *et al* [van de Beek1997] describe a maximum likelihood scheme employing a correlator, a moving sum and a peak detector used to synchronise a 5 MHz bandwidth (at 2 GHz) OFDM system over a modelled time-dispersive channel with a

maximum delay spread of $3\mu S$. A frame clock is generated at the receiver using the cyclic extension of OFDM frames.

Regarding carrier frequency synchronisation, a novel frequency synchronisation technique based on CAZAC training data transmitted using a single carrier frequency, was proposed by Lambrette, Speth and Meyr [Lambrette1997]. The properties of the CAZAC sequence are dependent on the linearity of the transmitter and receiver front-ends hence the transmitted sequence may be subject to distortion effects unrelated to the wireless channel effects. This may then result in an error in the channel equalisation stage. The effectiveness of this frequency synchronisation technique is based on the assumption that correct frame synchronisation has been achieved, however. Using the periodicity metric technique to determine the frame position and frequency offset, Speth, Classen *et al* [Speth1997] found that for coherent OFDM reception, frame alignment errors posed a significant threat to the resulting BER. This was deemed to be due to the resulting misalignment of the FFT windows. Since the channel estimation information was derived by interpolating between received known pilot symbols using a Wiener interpolation filter, FFT window misalignment would have resulted in the incorrect sub-carrier values being interpreted as the received versions of the known pilot symbols. This approach, which may be applied to DVB-T reception means that not all of the sub-carriers are used for data transmission as known pilots are required to be transmitted in order to maintain synchronisation.

Schmidl and Cox [Schmidl1996][Schmidl1997a][Schmidl1997b] also investigated the use of a training sequence of two OFDM training symbols to correct the carrier frequency offset and estimate the start of frame position. Each of the two training symbols has two identical halves. After passing through the channel, the carrier frequency offset may be calculated from the phase difference between two halves of one of these OFDM training symbols. Frame synchronisation may then be achieved using a correlation-based technique by exploiting the identical nature of the identical halves of the second OFDM training symbol. This technique enables frame synchronisation and carrier-frequency offset estimation to be achieved using two OFDM symbols transmitted at a specific time during the OFDM frame guard. In addition to removing the requirement for pilot sub-carrier transmission during the OFDM frame and thus improving the bandwidth-usage of the OFDM transceiver system, it is not necessary to have *a priori* knowledge of the pilot sub-

carrier values used. This technique is called ‘Maximum-Normalized Correlation timing and Carrier Frequency Offset (CFO) estimation’.

The subject of training data overhead for bursty transmission systems was addressed by Speth, Daecke and Meyr [Speth1998]. The findings presented (based on simulations and not on live data), indicated that the probability of a frequency synchronisation failure is independent of the SNR indicating that the failure probability may also be independent of flat-fading effects. The common correlation based symbol synchronization scheme shows that the performance will plateau and even decrease if the number of samples within the correlation window is too large. It was concluded that a higher degree of synchronisation accuracy may be possible using a single-carrier training sequence but this approach would negate the benefits of multi-carrier frequency diversity in terms of robust transceiver design.

Hsieh and Wei [Hsieh1999] proposed a guard-interval based low complexity frame synchronization and carrier frequency offset algorithm using the sign bits of the in-phase and quadrature components. The work presented in this particular paper is important due to the fact that it was also shown that frame synchronisation and frequency synchronisation are mutually dependent and therefore, must be estimated at the same time. This proposed algorithm, using the cyclic extension of OFDM symbols as the means to achieve synchronisation, extended the frequency acquisition range to half of the useful OFDM bandwidth. The low-complexity of the proposed technique was largely due to the inherent low-complexity of Maximum Likelihood (ML) Moving Average (MA) frame synchronisation technique and the channel model only consisted of an AWGN channel. In this case, the performance of the frame synchronisation algorithm would be expected to degrade significantly when tested using a wireless channel affected a multi-path and Doppler fading, impulsive and non-Gaussian coloured noise due to the false frame lock estimates.

Coulson [Coulson2001a] [Coulson2001b] presented an analysis of OFDM maximum likelihood synchronisation algorithms using a pilot symbol and provided analyses of methods used to determine suitable detection threshold values. Initial experimental results were derived from AWGN channel models and did not involve frequency-selective multipath fading channels affected by impulsive, coloured noise or adjacent channel interference. It was reported that a practical system would require a minimum SNR of 15

dB in order to minimise the possibility of synchronisation failure. Almenar, Abedi *et al* [Alenar2001] present a burst synchronization method for HIPERLAN/2 which is a preamble based OFDM wireless standard described in Chapter 2. This method was analysed under conditions of SNR, clipping and multipath effects. The proposed method is similar to the one proposed by Schmidl and Cox where the received signal is cross correlated with a delayed version of itself and the resultant signal is averaged and scaled by the mean signal power.

Seo, Kim, *et al* [Seo2002] approached optimum synchronization techniques using two OFDM training symbols. One training symbol is used for symbol timing recovery and the second symbol is used to estimate the carrier frequency offset. The proposed scheme relied on the phase difference between adjacent sub-carriers instead of the common sub-carrier correlation techniques being developed. This phase-difference based technique is highly susceptible to phase errors induced by noise, frequency-selective fading and Doppler fading so is not considered a robust enough scheme for the mobile fading channels envisaged for roaming reconfigurable radio devices in this thesis.

From these main examples and synchronisation scheme proposals, it is deduced that a low bandwidth overhead and robust OFDM frame synchronisation and carrier frequency offset estimation technique would be the preferred choice for an OFDM reconfigurable transceiver system. Using a non real-time OS means that quasi-realtime operation may be achieved, but this depends on the available processing power and the processing demands imposed on the system. As the OS used in the implementation platform is a non-real time OS, one of the main aims of the designer is therefore to ensure that the total number of clock cycles required to implement the complete reconfigurable radio design is less than the number of clock cycles that may occur before real-time constraints are breached. As a result, an ideal frame synchronisation and carrier-frequency offset estimation technique is one that also presents a low processing-power overhead to the reconfigurable transceiver.

7.4 Chosen Technique

The frame synchronisation and carrier-frequency offset estimation algorithm developed for the reconfigurable radio implementation presented in this thesis is based on the techniques originally developed by Schmidl and Cox. This synchronisation scheme presented combines the desired qualities of robustness, low-complexity and relatively small

processing-power and time overhead. Using this algorithm base, the techniques are developed further and adapted for use in a reconfigurable radio. The frame synchronisation technique involves transmitting one special OFDM pilot symbol composed of two identical halves in the time-domain. This OFDM pilot symbol is called a training symbol. Essentially, this training symbol is composed of two short identical OFDM symbols, which are transmitted in series to form one training symbol. Each half of this training symbol consists of a specific number of pilot symbols taken from points on a constellation diagram representing a high-order modulation scheme such as 16-QAM, 64-QAM or 256-QAM. The specific constellation points chosen correspond to signals with a high power in order to ensure that the receiver detects the pilot symbol. The receiver does not need to have *a priori* knowledge of the sub-carrier pilot symbols used.

This training symbol also functions as a means of estimating the Carrier Frequency Offset (CFO) that may exist between the transmitter and receiver. Frame and CFO estimation may be achieved using this technique. If the CFO is non-zero, then ICI and loss of orthogonality may occur. As each half of the training symbol is identical, a CFO may manifest itself as a phase difference between the corresponding pilot symbols in the first and second halves of the training symbol. If the wireless channel is approximated as being static for the duration of the training symbol, then if the CFO is zero, the two halves of the received training symbol are expected to be identical.

7.4.1 Operational Overview

From the point of view of the receiver, the frame synchronisation algorithm relies on searching the intercepted signal sequence for this training symbol denoting the start of a frame. The search method used views the incoming signal sequence as a series of possible training symbols (using the same structure as the expected training symbol) and correlates both halves of each possible training symbol. If both halves of this constructed symbol are identical (or almost identical), then the output of the correlator will register a significant increase. The index of the signal sequence at which the maximum correlation output is measured denotes the position of the training symbol in the intercepted signal sequence. As the training symbol may be transmitted at a specific instance during the frame guard specified by the transmission standard, the number of samples between that training symbol and the actual start of the frame is known, and is referred to as a fixed offset.

Therefore, the estimated index of the training symbol, offset by the known fixed offset, results in an estimate of the actual beginning of the frame.

7.4.2 Frame Synchronisation Algorithm

The method of OFDM generation and reception was already described in detail using Eqs. [2.1] to [2.10] in Chapter 2. Based on this original explanation, the generation of a baseband OFDM signal will be developed further in order to lay the foundations for the presentation of the frame synchronisation and CFO estimation technique. This approach differs from that of Schmidl and Cox's Maximum-Normalized Correlation timing and Carrier Frequency Offset (CFO) estimation technique as only one training symbol is created consisting of two identical halves in order to reduce the complexity and processing time required to estimate the location of the training symbol in a sequence of received signal values.

The two halves of the training symbol are made identical by inserting pseudo-random data symbols (pilot symbols) into the FFT array indices. Each complex-valued pilot symbol $c(k)$ has the same form as described by Eq. [2.4], reproduced here as Eq. [7.1]:

$$c(k) = A(k) \exp\left(\frac{2\pi k}{N}\right) = a(k) + jb(k) \quad [7.1]$$

where N in this case corresponds to the number of bits/symbol that the high-energy pseudo-random pilot symbol has been obtained from, $a(k)$ and $b(k)$ are the k^{th} In-Phase (I) and Quadrature (Q) values, respectively.

Regarding the first half of the training symbol, pseudo-random pilot symbols are inserted into the FFT array indices that designate the even-numbered frequencies and zeros are inserted into the FFT array indices designating the odd-numbered frequencies. The main difference between a normal OFDM symbol and a training symbol is that a normal OFDM symbol may contain non-zero data symbols on both the even and odd-numbered carrier frequencies. The pseudo-random pilot symbols are essentially complex-valued signal points on a QAM constellation diagram and the energy associated with this pilot symbols is high in order to ensure that the receiver detects the transmitted training symbol. For the second half of this training symbol, the same pseudo-random sequence of pilot symbols is

inserted into the even-numbered array indices and the odd-numbered array indices contain zeros. Using a similar approach as employed in Eq. [2.7], the FFT array has $N_{FFT(pilot)}$ bins, or array indices, where $N_{FFT(pilot)}$ denotes the size of the training symbol FFT.

This frame synchronisation algorithm is designed for a reconfigurable radio and one of the properties of this radio is that a variable length OFDM pilot symbol may be used to reduce or increase the complexity of the OFDM pilot symbol depending on the estimated channel SNR. In general the duration of the time-domain OFDM symbol is directly proportional to the FFT size. It is possible, therefore, that the OFDM pilot symbol FFT size may be different than the FFT size used for normal data transmission (i.e. N_{FFT}) in order to implement this variable length OFDM pilot symbol.

After the real pseudo-random pilot symbols have been inserted into the even-numbered FFT array indices for both halves of the entire training symbol, the complex-conjugate of these sub-carriers must be inserted into the FFT array indices corresponding to the Imaginary frequencies in order to be able to correctly generate the time-domain OFDM training symbols. This process is similar to the normal carrier-insertion procedure described in Chapter 2 except that the size of the FFT array size may differ as already stated.

Table 7.1 illustrates how one half of one OFDM training symbol FFT array may be specified using four pilot sub-carrier values. The completed OFDM training symbol is comprised of two of these half-symbols (or two short OFDM symbols). The sub-carrier notation denotes the carrier frequency, where 0 represents zero frequency (DC).

Table 7.1: Example of training symbol sub-carrier allocation using four pilot symbols.

Sub-carrier index	$c(k)$
-4	$8-8j$
-3	0
-2	$-8+8j$
-1	0
0	0
1	0
2	$-8-8j$
3	0
4	$8+8j$

This training symbol is then converted to a time-domain OFDM symbol and transmitted across a time-varying wireless communications channel using a similar process as described in Chapter 2. The received OFDM baseband signal is obtained by down-converting, digitising and performing an IFFT operation on the received OFDM signal sequence containing the training symbol. In order to locate the training symbol, the baseband digitised sequence of data symbols must be partitioned into data-symbol arrays.

The received signal is segmented into a sequence of possible training symbol arrays containing 18 signal points (each training symbol half contains 9 sub-carriers in this example). When the receiver detects that an actual OFDM training symbol has been received, then the first half of the symbol should be identical to the second half of the training symbol received $T/2$ seconds later, where T is the period of the OFDM symbol (i.e. $N_{FFT(pilot)}$ samples). Assume a slowly-varying communications channel, the effects of which may be considered static for the duration of the training symbol. A carrier frequency offset will result in a phase difference between the pilot sub-carrier value in one OFDM symbol half and the corresponding pilot sub-carrier value in the second OFDM symbol half. This phase difference may be approximated as follows:

$$\phi = \pi T \Delta f \quad [7.2]$$

where Δf is the carrier-frequency offset.

Consider a possible received training symbol comprising L complex-valued signal samples on one half of the symbol, where $L = N_{FFT(pilot)} / 2$. Let the sum of the pairs of products be denoted as:

$$\psi(k) = \sum_{m=0}^{L-1} r_{k+m+L} r_{k+m}^* \quad [7.3]$$

The received power for the second half-symbol may be described as:

$$P(k) = \sum_{m=0}^{L-1} |\psi(k)|^2 \quad [7.4]$$

As mentioned previously, the presence of a training symbol in a sequence of received signal values is expected to result in a significant increase in the output of the correlation-based frame synchronisation metric. Therefore, the location of the training symbol in the array of received signal values is estimated to be the array index that corresponds to the maximum value of $\hat{m}_{\max}$ where:

$$\hat{m}_{\max} = \arg \max_k (P(k)) \quad [7.5]$$

The training symbol may be transmitted during the frame guard between OFDM frames. The number of samples between the training symbol and the actual start of the frame is decided by the radio designer and an essential requirement in this case is that this sample offset must be known by the receiver in order to deduce the actual start of the frame. It is also possible to transmit the training symbol immediately before the guard interval of the first OFDM symbol in the OFDM frame. The drawback of this however, is that both the frame synchronisation and extraction of the OFDM symbols contained in the frame have to occur almost simultaneously and there exists a high possibility that the OFDM frame extraction procedure will lose a number of the frame samples due to processing time lags. By placing the training symbol in the middle of the frame guard, or closer to the end of the previous frame, the receiver has a specific processing time ‘window’ in which to detect the training symbol, estimate the start of the frame (which is offset from the estimated index of the training symbol by a known number of samples) and then invoke the OFDM frame extraction and demodulation algorithms.

In order to achieve fast frame acquisition, this frame synchronisation algorithm may be operating continuously. As a result, the output of Eq. [7.5] will vary continuously as the sequence of received signal values progresses through the synchronisation algorithm. In order to discriminate between noise, normal OFDM symbols and the training symbols, it is necessary to establish a threshold value, above which, it is decided that a training symbol has been received. A training symbol is deemed to have been received if the following condition is satisfied:

$$P(k) > threshold \quad [7.6]$$

7.4.3 Carrier Frequency Offset (CFO) Estimation

As mentioned previously, CFO estimation is important, particularly with OFDM-based systems, as carrier frequency misalignments between the transmitter and receiver may result in loss of received sub-carrier orthogonality. Eq. [7.2] shows that effect of a carrier frequency offset is that a phase difference will result between the first half of the received training symbol and second half of the training symbol, received $T/2$ seconds later. An estimate of the frequency difference may be obtained with the aid of the two received symbol halves using:

$$\Delta\hat{f} = \frac{\hat{\phi}}{\pi T} = \frac{\hat{\phi}}{\pi N_{FFT(pilot)}} \quad [7.7]$$

where $\hat{\phi} = \text{angle}(\psi(k))$ and $|\hat{\phi}| < \pi$.

CFO compensation may then be carried out by multiplying the received signal samples by Δc , where:

$$\Delta c = \exp\left(-j \frac{\hat{\phi} m}{N_{FFT(pilot)}}\right) \quad [7.8]$$

7.5 Implementation

Figure 7.3 is an illustration of the main processes involved in the creation of an OFDM frame. The OFDM modulator creates the OFDM symbols in parallel, where this representation of each OFDM symbol includes a pre-pended cyclic prefix guard interval. The blue areas in this figure describe the OFDM signal format following the stages indicated in the diagram. The OFDM frame construction block converts these OFDM symbols in parallel to a serial sequence of OFDM frames, where each frame contains N OFDM symbols (in Figure 7.3, $N=2$). Each OFDM frame is separated by a frame guard, which is simply a null signal of a specific duration. The training symbol insertion stage ‘inserts’ a training symbol in this frame guard interval. For illustrative purposes, this training symbol is shown in Figure 7.3 as being in the centre of the frame guard.

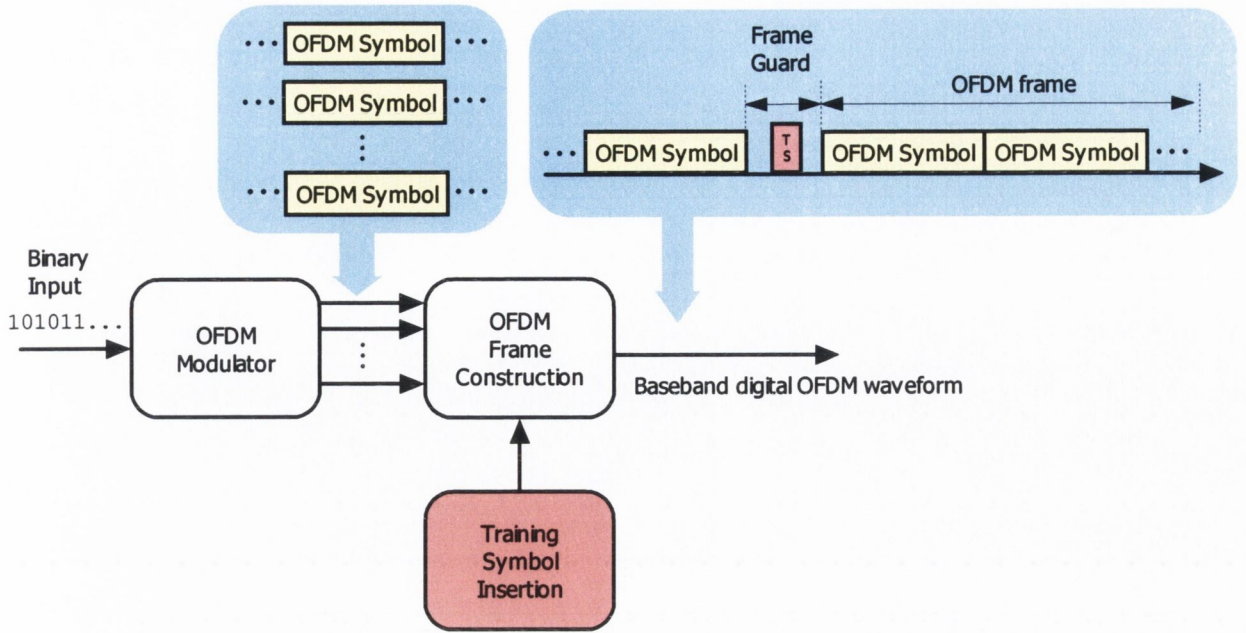


Figure 7.3: Overview of the training symbol insertion process.

The entire process was prototyped and simulated using MATLAB before integrating the required stages into a Radio Component and Control Logic routines for use with the IRIS system.

7.6 Evaluation

The IRIS implementation of this frame synchronisation and CFO estimation technique permits modification of the pilot FFT size, $N_{FFT(pilot)}$ using Control Logic in order to vary the complexity of the algorithm. In fact, it is possible to replace the frame synchronisation and CFO estimation technique with another synchronisation algorithm or invoke multiple different synchronisation technique in order to compare frame and frequency acquisition accuracy and processing power usage using the IRIS framework and Control Logic.

This frame synchronisation technique has been implemented using IRIS and the number of pilot sub-carriers can be varied depending on the estimated channel SNR. In other words, if the channel SNR is high, then a complex version of this frame synchronisation scheme (i.e. using the maximum number of pilot sub-carriers) may not be necessary. Therefore, in reasonably high SNR conditions, frame synchronisation can be achieved using fewer pilot sub-carriers thus reducing the algorithm complexity. The reconfigurable radio implementation of this scheme can be modified in terms of the pilot FFT size used, which

varies the duration of the pilot OFDM signal in the time domain. This allows frame acquisition to occur when the channel SNR is high, using a minimum-duration OFDM pilot symbol.

An illustrative example of how the output of the frame synchronisation algorithm performs with an received baseband OFDM waveform is shown in Figure 7.5. The incoming time-domain OFDM signal is shown in Figure 7.4 and consists of two training symbols followed by three OFDM frames. This OFDM signal is corrupted by AWGN and the SNR is 10 dB. Each OFDM frame consists of nine OFDM symbols and each OFDM symbol is preceded by a guard interval consisting of a null signal. This has been carried out for illustrative purposes, in order to visually indentify each OFDM symbol (i.e. normally one training symbol is used and each OFDM symbol is preceded by a cyclic prefix in order to minimise the possibility of loss of orthogonality due to multi-path fading channel effects). Each training symbol consists of 15 pilot sub-carriers and $N_{FFT(pilot)} = N_{FFT} = 1024$ in this example.

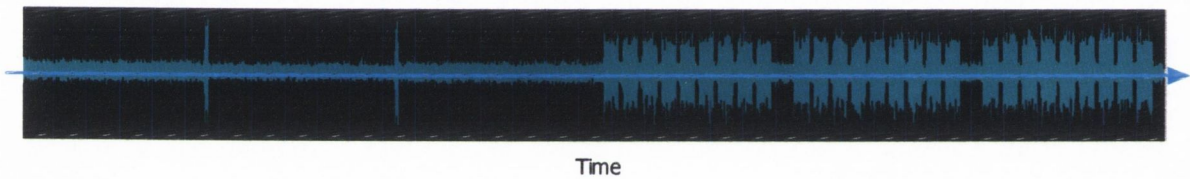


Figure 7.4: Time domain representation of three OFDM frames preceded by two training symbols (SNR=10 dB).

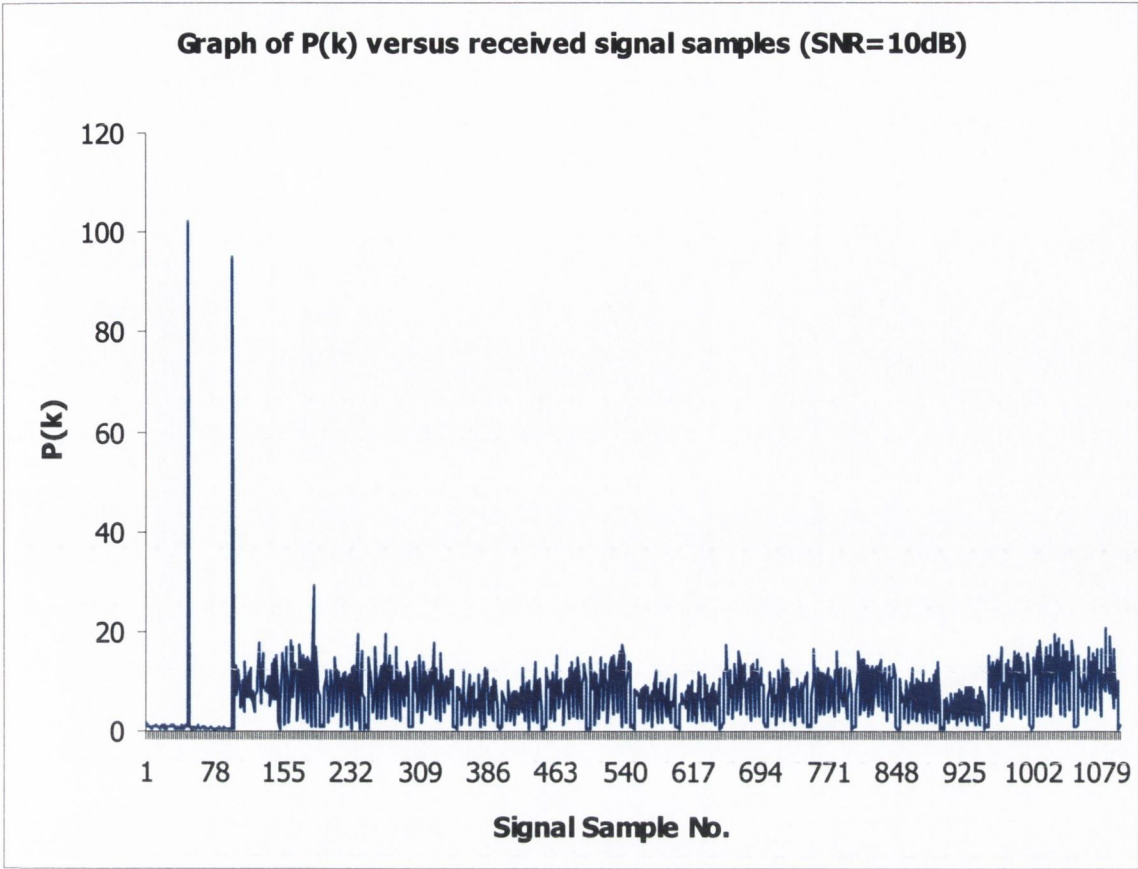


Figure 7.5: Graph of $P(k)$ versus received signal sample indices (SNR = 10 dB).

Figure 7.5 is a graph of $P(k)$ versus the signal sample index where the input signal is the OFDM signal sequence shown in Figure 7.4. Regarding Figure 7.5, it is clear that two distinct peaks (positives) are registered in the output of $P(k)$ when the training symbols encounter the frame synchronisation algorithm stage. During the rest of the frame guard, $P(k)$ is dramatically lower and for the duration of the three OFDM frames, the value of $P(k)$ varies due to the non-white OFDM spectrum resulting in random slightly correlated signal samples, although no significant peaks occur that will result in false positives.

The same time-domain OFDM signal is shown in Figure 7.6 except that for this case, the SNR is reduced to 0 dB. It is more difficult to visually distinguish between the OFDM frames, OFDM symbols and the training symbols. The performance of the frame synchronisation algorithm is also tested for this noisier signal case.

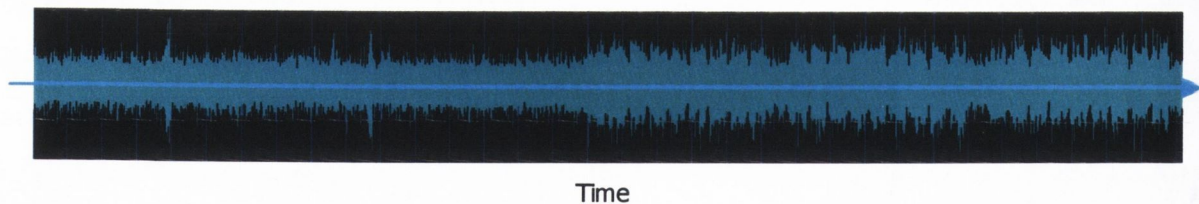


Figure 7.6: Time domain representation of three OFDM frames preceded by two training symbols (SNR= 0 dB).

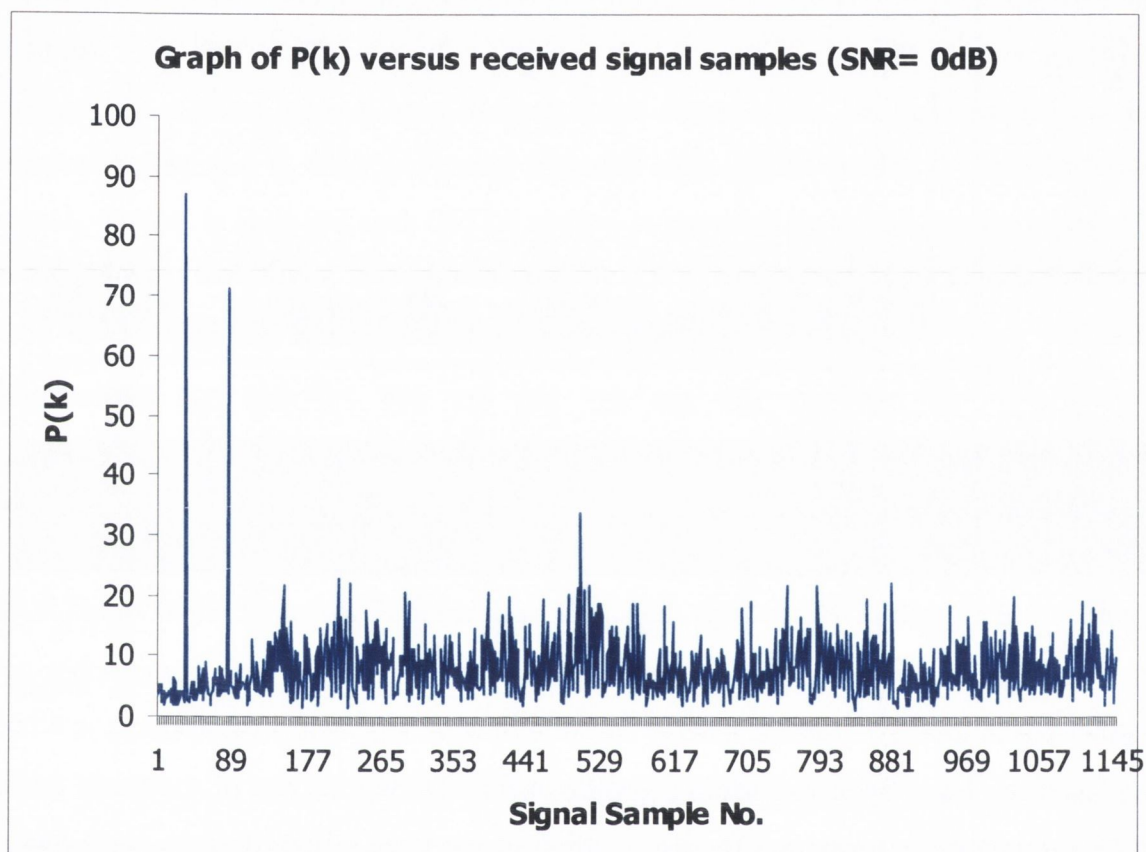


Figure 7.7: Graph of $P(k)$ versus received signal sample indices (SNR = 0 dB).

The graph of $P(k)$ versus the indices of the sequence of received baseband OFDM signal, where the SNR is 0 dB, is shown in Figure 7.7. Visually, the output of the frame synchronisation algorithm has degraded due to the reduced SNR. It is still possible to distinguish two peaks indicating positive identification of the two training symbols. It is important to point out that since the SNR for this case is 0 dB, it is reasonable to suggest that the BER of the received data for this case would be excessive, yet the start of the frame can still be estimated.

7.7 Conclusion

This chapter has presented a low-complexity frame synchronisation technique designed for GPP reconfigurable radio. A review of the main research work carried out in this area leads to an explanation of how the foundation for the proposed technique was established. The chosen frame synchronisation technique, based on the use of a training symbol, also facilitates the estimation of carrier frequency offsets that may exist between the receiver and transmitter. It was described how the proposed technique is capable of being reconfigured to vary the number of pilot sub-carriers in order to reduce the duration and complexity of the training symbol event further. Simulation results for AWGN channel SNR values of 10 dB and 0 dB showed that the start of an OFDM frame may be detected.

8

DYNAMIC OFDM

8.1 Introduction

This chapter presents a novel frequency-hopping OFDM technique, which is referred to in this thesis as Dynamic OFDM (DOFDM). The fixed-architecture OFDM modulator introduced in Chapter 2 is an example of a ‘static’ OFDM implementation as the operation or characteristics of the system may not be modified. In the case of DOFDM, the frequencies of the OFDM sub-carriers are chosen randomly from a set of possible sub-carrier frequencies at periodic intervals, either on a per-OFDM frame or a per-OFDM symbol basis. The number of sub-carriers used for transmission as part of the OFDM multiplex may be designated by the radio designer, but is limited by the size of the FFT. In the context of this thesis, DOFDM is understood to be an OFDM reconfigurable radio that employs frequency-hopping techniques to transmit and receive information instead of fixed, or ‘static’, sub-carrier frequencies. The frame synchronisation technique presented in the previous chapter is an essential part of the DOFDM technique. This chapter also shows that DOFDM has potential to increase the security of the transmitted information and to reduce the peak power of an OFDM transmission thus reducing the possibility of transmitter Power Amplifier (PA) saturation.

Section 8.2 outlines the context in which the proposed DOFDM technique will be implemented. The main advantages of the proposed technique are described in Section 8.3 followed by a more detailed description of a DOFDM reconfigurable radio implementation. Section 8.5 presents results demonstrating that the proposed scheme reduces the peak power of an OFDM transmission, as previously mentioned, and Section 8.6 concludes this chapter.

The work presented in this chapter has been published in Proceedings of the 2003 Institution of Electrical Engineers (IEE) Digital Signal Processing (DSP) Enabled Radio Colloquium [Nolan2003b].

8.2 Placing the Work in Context

Figure 7.1 illustrates the main signal-processing functions required for an implementation of an OFDM transceiver that was originally presented in Chapter 2. The wireless channels associated with the sub-carrier frequencies used in an OFDM system have propagation channel gain and noise profiles that are independent of each other. It is possible that certain frequencies may experience noise and interference that may result in temporary loss of the information that is transmitted using these frequencies. DOFDM increases the resilience of the transmitted OFDM symbols to these frequency-selective anomalies by reducing the length of time a sub-carrier is used to convey information. Dynamic reconfiguration of the OFDM modulator and demodulator is required to dynamically modify the sub-carriers used for transmission and demodulation. The IRIS reconfigurable radio is therefore an enabling platform for this scheme.

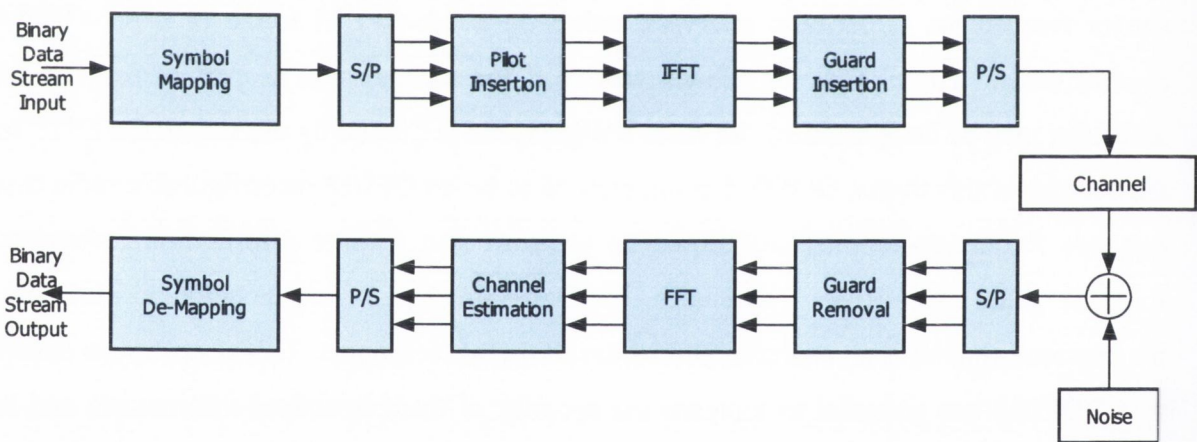


Figure 8.1: Block diagram of a basic OFDM transceiver.

The stages of the OFDM modulator and demodulator which are relevant to the DOFDM technique proposed in this chapter are highlighted in yellow in Figure 8.2. The Dynamic Sub-Carrier Selection stage designates the sub-carriers that will be used in each OFDM symbol multiplex. The Serial to Parallel (S/P) stage must then partition the incoming modulated sequence of data symbols into parallel sequences of data symbols according to the selected sub-carriers. The Sub-Carrier Extraction stage in the OFDM demodulator must also dynamically change in order to correctly extract the modulated information. Therefore the sub-carrier selection and extraction processes must be time-aligned, or

synchronised. The proposed scheme uses the detected frame synchronisation training symbol as the synchronisation point for the sub-carrier extraction process.

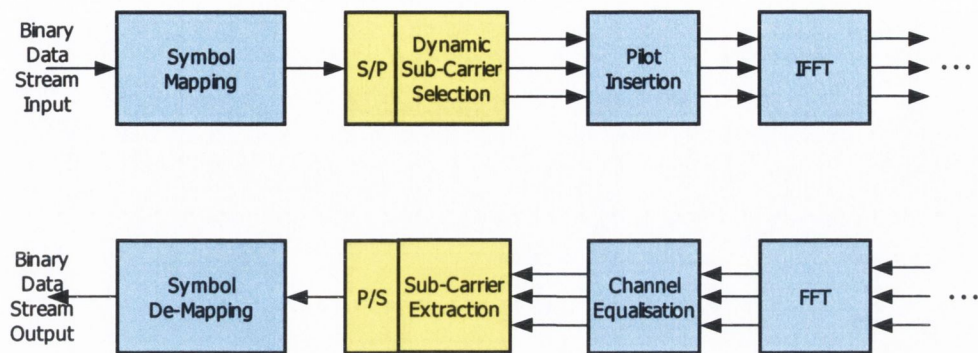


Figure 8.2: Block diagram of the OFDM transceiver highlighting the DOFDM-specific stages.

8.3 Advantages of DOFDM

The ability of an OFDM transceiver to dynamically change the sub-carriers used for transmission, either on a per-frame or a per-OFDM symbol basis means that four major advantages to the user and radio designer may be gained, namely:

- Increased Interference Immunity.
- Frequency Reuse.
- Information Security.
- Peak to Average Power Ratio (PAPR) Reduction.

8.3.1 Increased Interference Immunity

Firstly, through the use of frequency-hopping techniques, the frequency-diversity advantages of OFDM may be enhanced as each sub-carrier is used for a shorter amount of time. This means that the robustness of the transceiver against frequency-specific interference and noise may be increased.

8.3.2 Frequency Reuse

Secondly as well as increased interference immunity, frequency-hopping also means that two or more transceivers may possibly use the same frequency as each source may only be

transmitting on a carrier for a much shorter length of time than for non-frequency hopping OFDM schemes (i.e. ‘static’ OFDM). One result of this is that a base station may be able to support more users (OFDM devices) due to the reusability of carrier frequencies as each wireless device accessing a base station does not require exclusive access to a fixed set of carrier-frequencies for the duration of its operation. An example of an existing technology using this feature is the Flash-OFDM work being carried out by Flarion Technologies [Corson2001].

8.3.3 Information Security

Thirdly, the result of dynamically changing the sub-carriers on a per-frame or per-OFDM symbol basis means that the transmitted information is effectively ‘scrambled’. In order to successfully demodulate the transmitted data, an eavesdropper would require knowledge of the pseudo-random frequency-hopping sequences and the order in which each original parallel sub-set of data symbols was mapped to the sub-carriers. For example, if a PN-generated sequence of three sub-carrier indices is defined as {2,31,3}, then the first parallel sub-sequence, or data symbol is mapped to sub-carrier index 2, the second parallel data symbol is mapped to sub-carrier index 31, and the third parallel data symbol will be mapped to sub-carrier index 3. The receiver requires not only knowledge of the sub-carrier indices but also the order in which the data symbol to sub-carrier mapping was carried out. This is how the transmitted information is effectively scrambled thus making it more difficult for an eavesdropper to intercept and successfully demodulate the transmitted information provided that the original seed value used to generate the sets of PN sequences is not disclosed to the third party in question.

8.3.4 Peak to Average Power Ratio (PAPR) Reduction

An OFDM transmission with a high PAPR may result in non-linear amplification in the transmitter Power Amplifier (PA) stage if the dynamic range of the PA is not sufficient enough to amplify the peak OFDM signals properly. This can result in signal distortion, loss of orthogonality between sub-carriers and increased power consumption by the transmitter. One of the challenges of OFDM transmission systems are therefore managing the high Peak-to-Average Power Ratio (PAPR) [VanDerOuderaa1988][Wu1995][Ngajikin2003] of OFDM signals. Signal distortion occurs when the signal is not amplified linearly across the frequency range in use and more

importantly, the amplifier may become overloaded due to a high signal level [Gifford2002]. Large peaks in the power of the signal may attempt to exceed the amplifier saturation region and result in distortion of the signal and possible unwanted spurious emissions. Therefore, the challenge is to amplify the base-band OFDM signal without distorting the signal. The non-linear effects of a power amplifier when subjected to a input signal with a high peak PAPR result in the transmitted signal being subjected to two main distortion products, Inter-Modulation Distortion (IMD) and harmonics. The large number of sub-carriers used in some OFDM systems result in many instances of these distortion products. The topic of PAPR has been the topic of a significant amount of research material and proposed techniques for reducing the PAPR of OFDM transmissions include Tellado and Cioffi's [Tellado1998] technique of using non-data-carrying sub-carriers (i.e. data symbols with zero value) to reduce the PAPR by inserting data-symbols in the non-data-bearing sub-carriers that actually counter the adjacent high power data symbols. These special symbols used to reduce the peak power of the OFDM waveform are not demodulated as data-bearing symbols at the receiver. Other proposed PAPR-reduction techniques include amplitude limiting [Dinur1998] [Wulich1999] and data symbol coding techniques [Kamerman1994]. Wilkinson [Wilkinson1995] proposed a block-coding technique that reduced the PAPR of an eight-carrier OFDM transmission by 50% using a 7/8 code rate and Narahashi [Narahashi1994] and Tarokh [Tarokh1999] examined a method of setting the initial phases of the sub-carriers that would result in a minimum PAPR of approximately 3.01 dB.

8.4 DOFDM Implementation

The main part of a reconfigurable radio implementation of DOFDM using the IRIS system is the pseudo-random, or pseudo-noise (PN) Sequence Generation/Re-generation and Dynamic Sub-Carrier Selection/Extraction stages highlighted in yellow in Figure 8.3. Sub-carrier indices are obtained by randomly generating N_{SC} unique sub-carrier indices from a total number of N_{FFT} sub-carriers, where N_{FFT} is the FFT size and $0 < N_{SC} \leq N_{FFT}$. The chip-rate of the frequency-hopping 'spreading code' is therefore equal to the number of sub-carriers required, (i.e. N_{SC}).

The serial-to-parallel (S/P) conversion stage converts the serial modulated information sequence into N_{SC} parallel sub-sets of data symbols. Following this stage, the dynamic

sub-carrier allocation stage assigns the N_{SC} parallel data symbol subsets of modulated information requiring transmission to the chosen N_{SC} sub-carriers. The criterion for selecting the sub-carriers (i.e. the spreading code) is described later in this chapter. With knowledge of the spreading code, the receiver may then extract and re-assemble the intercepted information using the sub-carrier extraction stage to obtain the information transmitted by the N_{SC} sub-carriers. The parallel-to-serial (P/S) stage converts the N_{SC} recovered modulated complex-valued data symbol subsets back to a serial sequence before passing the re-assembled signal sequence to the symbol de-mapping stage. This symbol de-mapping stage demodulates the signal sequence and produces a binary sequence which ideally is the same as the originally transmitted binary sequence.

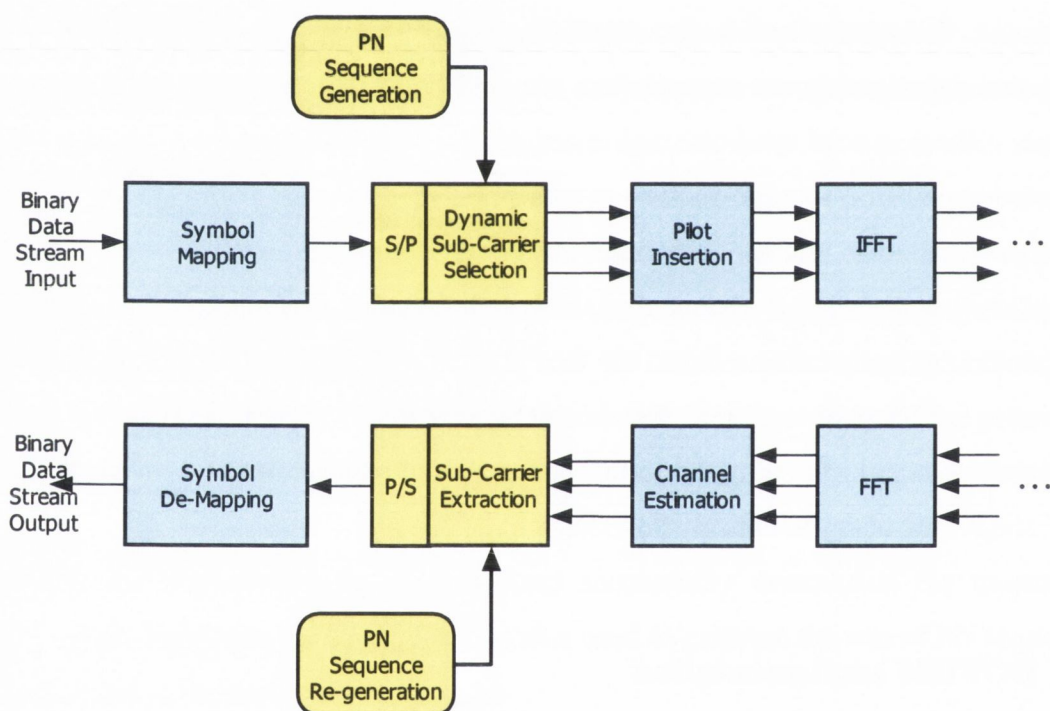


Figure 8.3: Block diagram of a DOFDM transceiver implementation.

8.4.1 Receiver Synchronisation

The receiver must have knowledge of the generated sub-carrier sequences and the hopping patterns. In this implementation, a new set of sub-carrier indices is required for each OFDM frame. This means that the receiver must be capable of obtaining the sub-carrier sequences being used and establish synchronisation with the transmitter in order to successfully extract and demodulate the transmitted information. For the purposes of

DOFDM synchronisation, the frame synchronisation technique presented in the previous chapter is also used to facilitate DOFDM synchronisation.

The means of generating the random sets of sub-carrier indices is to use a pseudo-random number generating function obtained from the Intel[®] Signal Processing Library [IntelSPL]. The `nspwRandUni()` function, illustrated in Figure 8.3 as the PN Sequence Generation and PN Sequence Re-generation block, is used to generate uniformly distributed pseudo-random numbers (representing sub-carrier indices) one at a time. In order to ensure that each of generated sub-carrier indices within the desired set is unique, the latest sub-carrier index is compared to the existing set of indices. If this sub-carrier index does not already exist in the set, then it is added to the set, or otherwise discarded.

The output of this function is dependent on an initial seed value. One of the properties of the `nspwRandUni()` function is that if the same initial seed value is used for two or more instances of this function, then the exact same sets of PN sequences may be generated. In other words, two or more remote instances of this function may generate identical sets of sub-carrier indices if the exact same seed value is used. This presents one solution for the problem of how to notify the receiver of which sub-carrier indices are being used by ensuring the remote transceivers:

- Possess identical copies of the P sub-carrier indices (spreading code) sets.
- Commence cycling through the P sets of sub-carrier indices at the exact same time.

In this implementation, P sets of N_{sc} sub-carrier indices are generated during the initialisation stage of the local and remote transceivers. The initial seed value is known to all of the devices beforehand using the `//@default` declaration representing the seed value in the IRIS Radio Component C++ header file. As discussed in Chapter 6, by using the default value declaration, the parameters of every instance of a Radio Component will have the same default seed value unless the remote user specifies a different seed value in the IRIS XML configuration file.

The number of sets, P , determines how many sets of N_{sc} sub-carriers are generated. In Figure 8.4, three sets of sub-carrier indices ($P=3$) are generated. The FFT size, N_{FFT} , is 8

and in this example, there are three OFDM symbols per frame. As a result, three sub-carriers out of the set of 8 are chosen using the `nspwRandUni()` function for each of the P sub-carrier sets. For each OFDM symbol numbered one to three, the corresponding sub-carrier set is used to denote which sub-carriers transmit the information. This is shown in the diagram as the red blocks indicating a chosen sub-carrier for each of the OFDM symbols. During the operation of the radio, when the final set of sub-carrier indices is used, the radio application returns to the first set of sub-carrier indices and begins the cycle when a new received frame is received and detected using the frame synchronisation training symbol. This continuously looping operation pattern is repeated until the radio is halted.

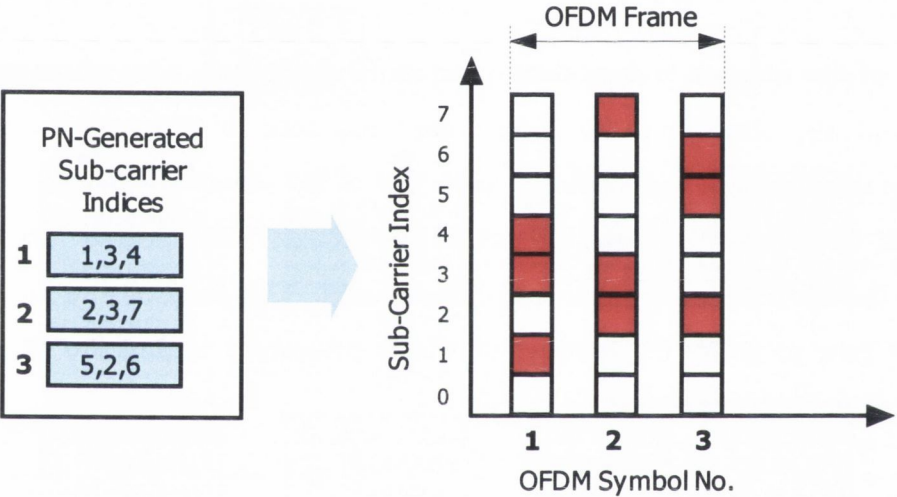


Figure 8.4: 3 sets of PN-generated sub-carrier indices for 3 OFDM symbols using a max FFT size of 8 bins.

In order to ensure that the remote reconfigurable transceiver(s) commence cycling through the P sets of PN-generated sub-carrier indices, the DOFDM scheme may be combined with the frame synchronisation scheme discussed in the previous chapter. If $P = N_{SYM}$, where N_{SYM} denotes the number of OFDM symbols contained in an OFDM frame, then each OFDM frame may correspond to one complete cycle of set of P sub-carrier indices. This means that each OFDM symbol in an OFDM frame may be numbered in sequence and assigned a particular sequence of sub-carrier indices from the set of P generated sub-carrier indices. When the local transceiver (the host) commences the transmission of an OFDM frame, the sub-carriers used in each OFDM symbol in the frame are derived by cycling through the set of P sub-carrier indices. Upon detection of a training symbol, the remote transceiver(s) may then begin cycling through the generated set of P sub-carrier indices,

starting with the first sub-carrier sequence specified in this set of P sub-carrier indices. This pattern of operation may then be repeated each time a training symbol denoting the start of an OFDM frame is received.

8.5 Evaluation

Using the IRIS system, tests relating to the measurements of the PAPR for the static and dynamic OFDM cases were carried out by creating an IRIS reconfigurable radio application comprising of binary sequence generator (information source), PN sequence generator, OFDM modulator and file writer (for analysis) Radio Components. The basic IRIS configuration block diagram is shown in Figure 8.5. This basic test system exists purely in the software domain.

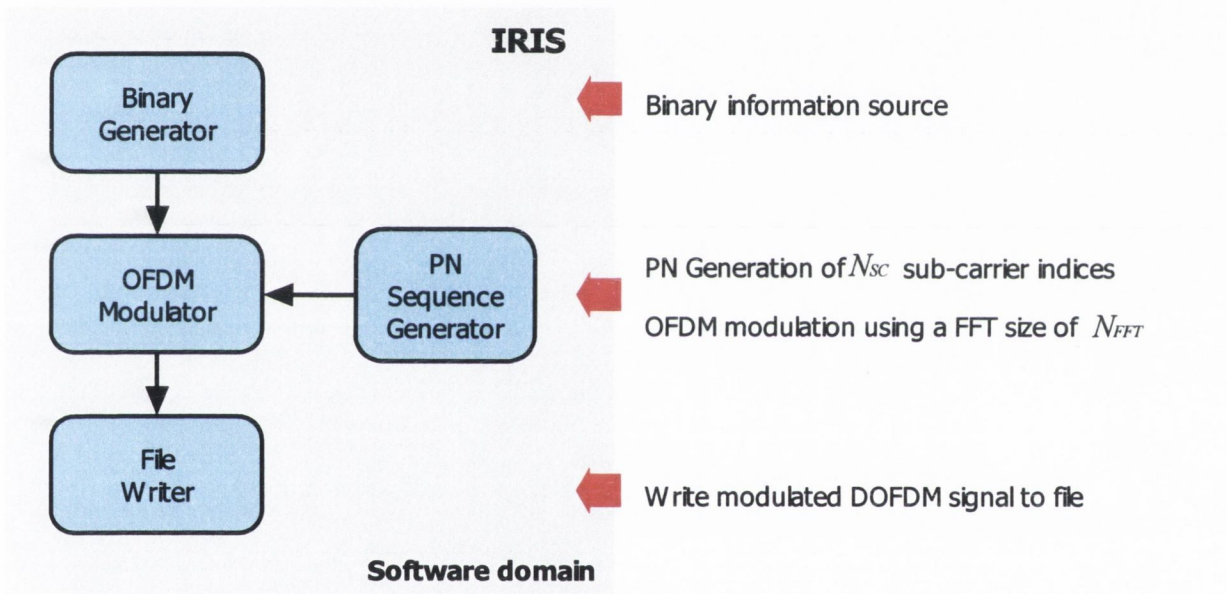


Figure 8.5: Block diagram of the IRIS DOFDM test modulator.

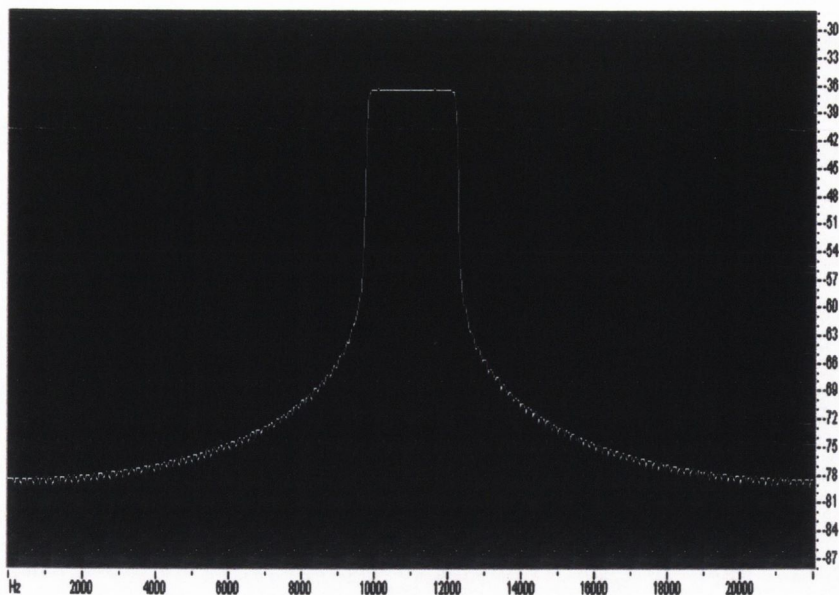


Figure 8.6: PSD of a static 54-sub-carrier OFDM signal.

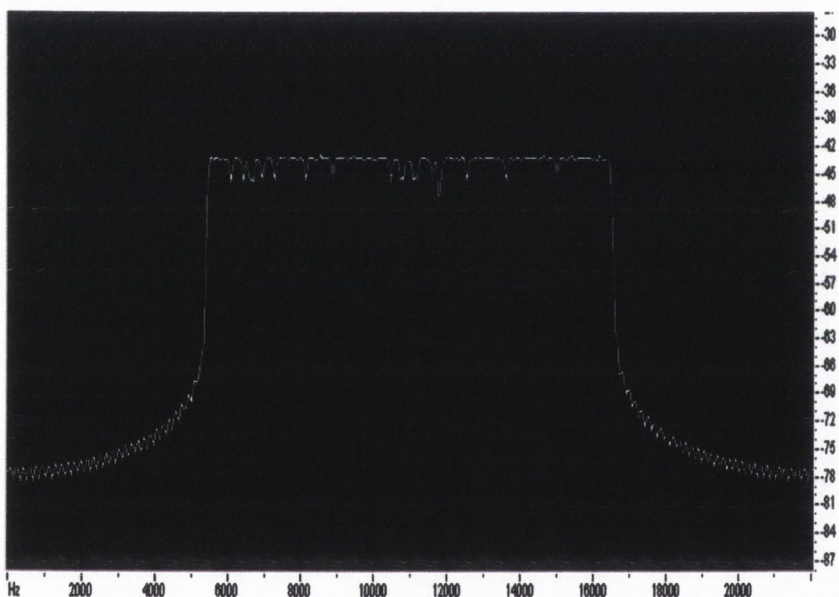


Figure 8.7: PSD of a dynamic 54-sub-carrier OFDM signal.

Examples of how the PSD of a static and dynamic OFDM implementation vary are shown in Figure 8.6 and Figure 8.7. In both cases, $N_{SC} = 54$, $N_{FFT} = 128$ and $N_{SYM} = 10$. Referring to Figure 8.6, the peak OFDM, averaged over N_{SYM} symbols is -34.6 dBm. In Figure 8.7, the peak value of the averaged PSD of the DOFDM implementation is -43 dBm and the bandwidth of the transmitted signal has spread to occupy the entire bandwidth that is possible using N_{FFT} sub-carriers.

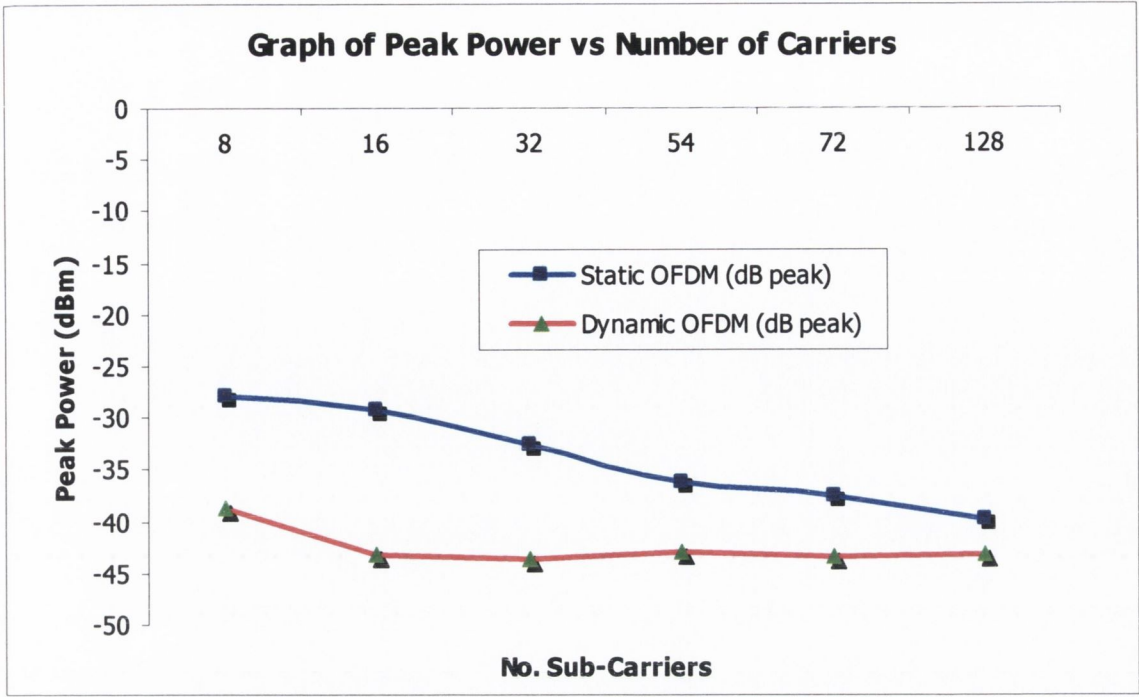


Figure 8.8: Graph of the peak power versus number of sub-carriers for fixed sub-carrier and DOFDM using IRIS.

Figure 8.8 is a graph of the peak power versus the number of sub-carriers, N_{SC} used for transmission out of a total number of N_{FFT} possible sub-carriers and averaged over 10 OFDM symbols and $P = 10$. In this example, $N_{FFT} = 128$ and N_{SC} is varied from 8 sub-carriers to the maximum of N_{FFT} sub-carriers (in this case). The blue line on this graph is the peak power of the OFDM waveform for a static sub-carrier configuration, (i.e. the same N_{SC} sub-carrier indices are used) for each of the N_{SC} allocations and the red line is the graph of the peak power of the OFDM waveform when N_{SC} sub-carriers out of the total set of N_{FFT} possible sub-carriers are chosen for each OFDM symbol.

It was found that by the DOFDM implementation significantly reduces the peak OFDM power when $N_{SC} < N_{FFT}$. In fact, the minimum value of the peak DOFDM power occurred for the $N_{SC} = 8$ sub-carriers case, with a measurement of almost -39 dBm while the peak power of the static OFDM implementation using the same number of sub-carriers was measured as -28 dBm. Using DOFDM in this case reduces the peak OFDM power by over 10 dB. The peak power of the DOFDM signal for the cases where $N_{SC} = \{16, 32, 54, 72, 128\}$ remains approximately constant at -43 dBm.

As well as reducing the peak power of the OFDM signal, the dynamic OFDM approach means that the amplitude of the DOFDM signal is effectively limited as discussed above and shown in Figure 8.8. This means that a RF front-end may be designed to offer linear amplification for signals with this peak power. A further implication of this is that a RF front-end with a lower linear-amplification specification may be used rather than a RF front-end capable of linear operation (a higher dynamic range) at higher peak OFDM powers in the case of the static OFDM transmissions. This means that the cost of the RF front-end may possibly be reduced.

8.6 Conclusion

This chapter has presented a technique that may be used to increase the robustness of OFDM when used in a wireless environment subject to interference, noise and frequency-selective fading. The proposed DOFDM technique uses the frame synchronisation training symbol described in the previous chapter as a means of synchronising the frequency-hopping sequences. It has been shown that this proposed DOFDM technique also acts a data scrambling technique, where the sub-carriers change on a per OFDM symbol or OFDM frame basis thus may also be used to increase the security of the transmitted information. Results presented in this chapter show that the peak power of the transmitted OFDM signal may be significantly reduced thus reducing the possibility of non-linear transmitter PA operation, signal distortion and spurious emissions. The IRIS reconfigurable radio platform is the enabling technology for this scheme and is another example of how reconfigurable radio may be used to improve OFDM transmission technology even further.

9

SUB-CARRIER ALLOCATION

9.1 Introduction

This chapter deals with a proposed sub-carrier allocation technique that may be used to dynamically allocate frequency segments to one or more users of a single reconfigurable OFDM transceiver. As the capabilities and signal-processing power of software radio/reconfigurable radio technology improves rapidly, spectrum allocation is now a feasible option for improving communications over increasingly crowded frequency bands. Sub-carrier allocation is an intelligent means of selectively assigning sub-carriers to transmit one or more sources of information based on the quality of the wireless channel associated with these carrier frequencies and/or the data rate required by the user(s) [Larsson1996]. Regarding the block diagram of the OFDM reconfigurable transceiver referred to in this chapter, the reconfigurable radio components directly related to the topic of sub-carrier allocation in this thesis are highlighted in yellow in Figure 9.1.

The work presented in this chapter has been published in Proceedings of the 4th Software Defined Radio Forum (SDRForum) Technical Conference in 2004 [Nolan2004].

9.2 Placing the Work in Context

Using the IRIS reconfigurable transceiver, which is common to all users, several independent sources of information may be transmitted to one or more other transceivers in the form of an OFDM multiplex. Sub-carrier allocation is developed from the dynamic OFDM (DOFDM) technique discussed in the previous section. The IRIS reconfigurable radio system and reconfiguration techniques are required to perform the dynamic conversion of one or more sources of binary data into modulated OFDM waveforms. The process of modulation and frequency-translation to one or more carrier frequencies is

referred to in this thesis as ‘dynamic data-symbol to sub-carrier mapping’. This means that the IRIS reconfigurable radio framework facilitates the dynamic data-symbol to sub-carrier mapping and de-mapping (sub-carrier demodulation) procedures, which may occur on either a per-OFDM symbol or per-OFDM frame basis in both the transmitter and receiver stages of the reconfigurable OFDM transceiver(s).

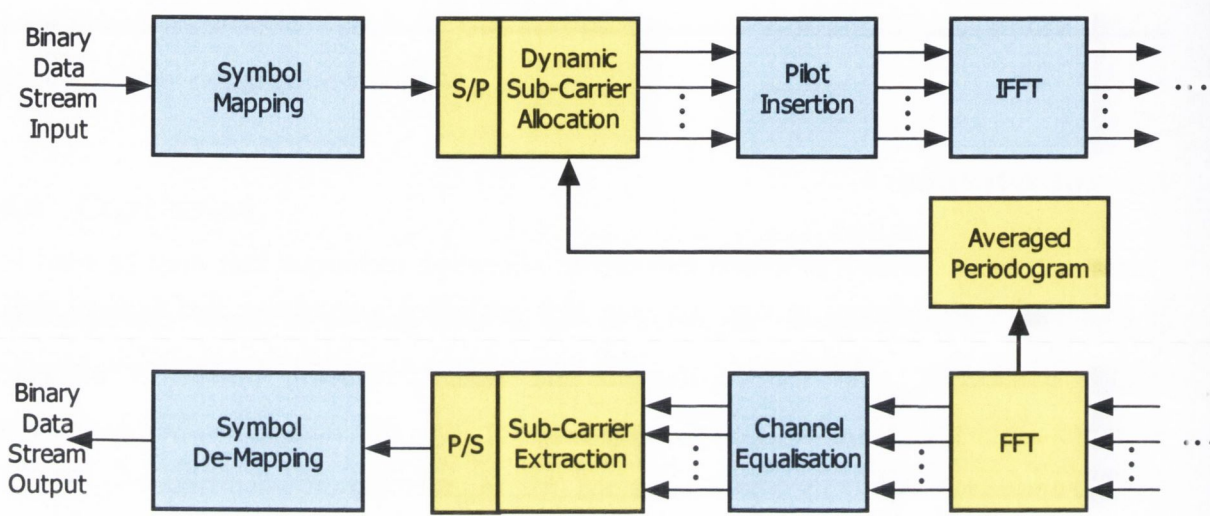


Figure 9.1: Block diagram of the basic reconfigurable OFDM transceiver sub-carrier allocation components of interest.

9.3 Review of Techniques

This section provides an overview of how OFDM evolved from fixed or ‘static’ sub-carrier usage to more dynamic frequency allocation techniques, capable of being implemented using the IRIS reconfigurable radio platform.

One of main differences between the frequency-hopping scheme described in the previous chapter and sub-carrier allocation schemes is that sub-carriers are selected based on the channel gain and level of noise and/or interference measured on the channel. A feedback mechanism from the receiver is therefore an essential part of the frequency allocation process. This means that sub-carriers may not be used if the level of interference and/or noise measured on any particular sub-carrier, or group thereof, may result in the unrecoverable loss of the transmitted information. Secondly, the emphasis is on maximising the number of sub-carriers that may be used for information transmission with a high probability that the transmitted information will be successfully received by the

remote reconfigurable OFDM transceivers. In order to intelligently assign sub-carriers for transmission of one or more information sources, a means of gauging the current level of spectral-activity on the particular band of interest is required. The method of ascertaining the amount of activity on a frequency band in the technique proposed in this thesis is to use determine the estimated PSD of the intercepted signal.

A dynamic sub-carrier allocation scheme based on a 'water-filling' concept [Cover1991] [Viswanath2001] and adaptive modulation [Harper1974] [Souryal2001] was proposed by Chow, Cioffi *et al* [Chow1995] that was reported to increase spectral efficiency thanks to the inherent multi-user diversity properties of OFDM multiple-access [Czylwik1997]. Chow concludes that although a sub-optimal allocation scheme is used to reduce the algorithmic complexity, simulation results show that up to 130% of capacity gain over that attainable using a non-dynamic frequency allocation Time Division Multiple Access (TDMA) scheme is achievable. Nakahara, Moriyama *et al* [Nakahara1996] concluded that OFDM as a means of more efficiently using the available bandwidth for a Digital Sound Broadcasting (DSB) system than transmission systems based on Single Frequency Networks (SFN). A comparison of fixed, random and interference-avoidance frequency resource allocation techniques for OFDM-multiple access was reported by Wahlqvist, and Olofsson [Wahlqvist1997] in 1997. The main conclusion is that dynamic resource allocation is better than fixed sub-carrier allocation. In 1999, Wong and Cheng [Wong1999a] [Wong1999b] showed that the inherent spectral efficiency of OFDM may be increased further by integrating adaptive sub-carrier, data symbol and power-allocation techniques into the standard OFDM transceiver system. In addition, this proposal facilitated shared access to an OFDM transceiver by enabling information from multiple users to be multiplexed as a sequence of OFDM symbols.

Research findings by Rhee and Cioffi [Rhee2000] showed that the channel characteristics for each sub-carrier in an OFDM system may be viewed as being mutually independent as common effects such as path-loss, multi-path fading and noise are generally frequency-dependent anomalies. Of course, as OFDM uses closely spaced sub-carriers, the channel perturbations encountered on a sub-carrier may also affect adjacent sub-carriers. One of their conclusions is that sub-carrier allocation may be used to attain a higher data rate through a wireless channel.

More recent studies of how to improve the capacity of multi-user transceiver systems have favoured the characteristics of OFDM and a means of dividing a spectrum segment between users. Alen, Madhukumar and Chin [Alen2003] proposed a frequency allocation scheme that divides the available bandwidth into equal partitions and ordered in terms of the channel gain (or 'usage' factor). The main criterion for this frequency allocation is the estimated gain, or usability of the channel. A low usage factor in this case refers to the gain of sub-carriers that have been estimated as experiencing severe fading effects and thus may not be capable of successfully conveying information to the destination transceiver(s). The frequency band partitions are ranked in order of increasing usage factors and assigned to users based on the usage factor and demand for each partition. Alen's proposal also incorporates adaptive modulation techniques such as employing a modulation scheme with a high bit-to-symbol ratio on frequency partitions with a high usage factor. Simulation results shown that the BER was reduced and channel capacity was increased by using this scheme however, the allocation plan was updated only after 1000 OFDM symbols. Although no details of the duration of the simulated OFDM symbol were given, the usage-factors of each partition vary with time. This means that there are potentially significant differences between the initial usage factors and those measured at the end of this OFDM symbol sequence. Zhen, Geqing *et al* [Zhen2003] exploits the independent nature of OFDM sub-carriers to propose an algorithm for optimal sub-carrier and power allocation in wideband OFDM transceivers. This proposal is also based on the data rate required by the user and the estimated channel gain and one of the conclusions is that by using such a resource allocation scheme, the power required to transmit information may be reduced. A distributed power and adaptive modulation algorithm proposed for dynamic sub-carrier allocation was evaluated by Lei and Zhang [Lei2004]. Simulation results for a 64 sub-carrier-based allocation scenario show that adaptive OFDM enables the number of transmitted bits per sub-carrier to be increased over a typical fixed modulation and sub-carrier OFDM scheme.

9.4 Power Spectral Density (PSD)

The Power Spectral Density (PSD) of an information signal may be used to determine RF signal power density over a specific frequency range. This information can be displayed using spectrum analyser or used to influence the operation of a reconfigurable radio. Using the PSD, it is possible to class independent orthogonal sub-carriers in terms of their associated channel quality and 'usage' prospects. The sub-carrier allocation algorithm can

then use a modulation scheme with a high bit-to-symbol ratio on the higher quality channels and conversely, use a more robust modulation scheme with a lower bit-to-symbol ratio on the lower quality channels. For the purposes of the sub-carrier allocation technique proposed in this thesis, the PSD information is used to avoid sub-carriers with a PSD greater than a specific threshold. The reasoning for this is that if a sub-carrier, which is experiencing a strong interfering signal or a high level of noise, is used to transmit information as part of the OFDM multiplex, there is a significantly high possibility that this transmitted information will not be detected by the intended receiver.

The PSD of an intercepted signal may be obtained by using the Discrete Fourier Transform (DFT) or the Fast Fourier Transform (FFT) [Cooley1965][Oppenheim1999]. As the FFT is already core to the OFDM transceiver already, as explained in Chapter 2, and essential to sub-carrier de-multiplexing process in the OFDM receiver, it is therefore possible to obtain the one-sided PSD of an intercepted signal without significantly contributing to the overall complexity of a reconfigurable OFDM transceiver.

An intercepted signal may be interpreted as existing as a time-varying discrete-time random signal. An averaged estimate of the time-dependent power density spectrum may be obtained using an averaged periodogram [Bartlett1953] [Welch1967], which is the time-averaged unbiased estimate of the power spectrum. This power spectrum is a smoothed and noise-reduced version of the PSD of the original input signal sequence. Time-averaging is used in order to prevent the sub-carrier allocation scheme discarding sub-carriers if the instantaneous PSD of the input signal was used in the sub-carrier allocation scheme. In other words, a short burst of impulse noise or interference may not necessarily mean that the sub-carrier is occupied by one or more other transmissions over the duration of the lifetime of the current sub-carrier allocation period. A time-averaged PSD is there a much better estimate of the amount of spectral activity occurring on a sub-carrier as very short duration impulses are smoothed out. It is important to note that if the intercepted time-domain signal is not zero-mean, a large power spectrum component will occur at zero frequency. This can reduce the dynamic range of the periodogram and decrease the amount of available information relating to each sub-carrier of interest [Oppenheim1999].

9.5 Chosen Technique

The method chosen to obtain the time-averaged PSD of the received signal based on Welch's periodogram technique. Consequently, the frequency-domain representation of

the received signal is required. The FFT stage of the reconfigurable OFDM transceiver plays a key role in this technique therefore, as this stage segments the intercepted, down-converted and digitised received signal as part of the OFDM receiver process already. Using this information, it is a relatively straightforward process to obtain the time-averaged PSD of the frequency band of interest.

Each signal segment is windowed using a windowing sequence denoted as $w(k)$, $k = 0, \dots, N_{FFT} - 1$. For each segment of length N_{FFT} , the windowed signal sequence $z(k)$, $k = 0, \dots, N_{FFT} - 1$ is formed, where $z(k)$ is defined as:

$$z(k) = r(k) \times w(k) \quad [9.1]$$

and $r(k)$ is expressed by Eq. [2.10].

A group of signal segments that will be used to obtain the time-averaged periodogram is formed from each of the L signal segments, where L is the number of segments that will be used to obtain the averaged estimated PSD. This set may be expressed as:

$$z_p(k) = r_p(k) \times w_p(k), \quad p = 0, \dots, L - 1 \quad [9.2]$$

Using the FFT procedure originally expressed in Eq. [2.11], the FFT of each signal segment may be expressed as:

$$A_p(n) = \frac{1}{N_{FFT}} \sum_{k=0}^{N_{FFT}-1} z_p(k) \exp\left\{-j2\pi \frac{nk}{N_{FFT}}\right\}, \quad p = 0, \dots, L - 1 \quad [9.3]$$

and $k = 0, \dots, N_{FFT} - 1$.

The averaged Fourier transform is a two-sided bandwidth of the input signal segments (i.e. the averaged Fourier transform of the positive and negative frequencies, in the range $\{-F_s/2, F_s/2\}$, corresponding to the FFT array values in the range $\{0, N_{FFT} - 1\}$, where F_s is the sampling frequency), has been obtained using the FFT process. These positive and negative frequencies, which are mirror images of each other, are contained in the complex-valued FFT array of size N_{FFT} . The positive frequencies are of interest (i.e. from zero Hz

to $F_s/2$) in this technique. Zero frequency is represented in the FFT array as the signal sample contained in array index $N_{FFT}/2$. The signal sample corresponding to $F_s/2$ is contained in array index $N_{FFT} - 1$. Therefore in order to extract the frequency range of interest, the averaged Fourier transform array denoted by A_p , $p = 0, \dots, L-1$ is reduced to $N_{FFT}/2$ samples by eliminating the unwanted negative frequencies.

At this stage, the time-averaged periodogram containing $N_{FFT}/2$ samples representing the frequency range from zero frequency to $F_s/2$. The final stage of the time-averaged periodogram derivation involves estimating the power of the signals contained in this frequency range.

The L modified periodograms, $I_p(f_n)$, representing the time-averaged PSD of the input signal may be expressed as:

$$I_p(f_k) = \frac{N_{FFT}}{U} |A_p(k)|^2, \quad p = 0, 1, \dots, L-1 \quad [9.4]$$

$$\text{where } f_k = \frac{k}{N_{FFT}}, \quad k = 0, \dots, N_{FFT}/2 \quad [9.5]$$

$$\text{and } U = \frac{1}{N_{FFT}} \sum_{k=0}^{N_{FFT}-1} W^2(k) \quad [9.6]$$

where $W(k)$ is the k^{th} element of the Fourier transform of the windowing sequence w .

Finally, the estimate of the PSD of the input signal $\hat{P}(f_k)$ may be obtained in the following manner:

$$\hat{P}(f_k) = \frac{1}{L} \sum_{p=0}^{L-1} I_p(f_k) \quad [9.7]$$

Using this analysis of the time-averaged PSD, or periodogram, the sub-carrier allocation scheme may decide to use each particular sub-carrier depending on the level of activity monitored on each particular sub-carrier.

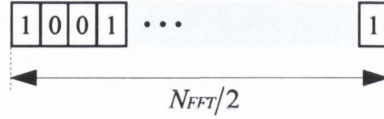


Figure 9.2: Channel mask example.

The sub-carrier allocation algorithm represents the $N_{FFT} / 2$ sub-carriers that may be used for transmission as part of the OFDM multiplex as an array of $N_{FFT} / 2$ binary values. This binary value array forms what is referred to in this thesis as a channel mask. An example of this channel mask is shown in Figure 9.2. If a sub-carrier is deemed ‘usable’, the binary value corresponding to that sub-carrier is binary 1. If a sub-carrier is deemed ‘unusable’, then the binary value associated with that sub-carrier is binary 0. Referring to the example shown in Figure 9.2, this means that the sub-carriers associated with index 0 and index 3 are useable while the sub-carriers associated with the channel mask array indices 1 and 2 are designated as being unusable.

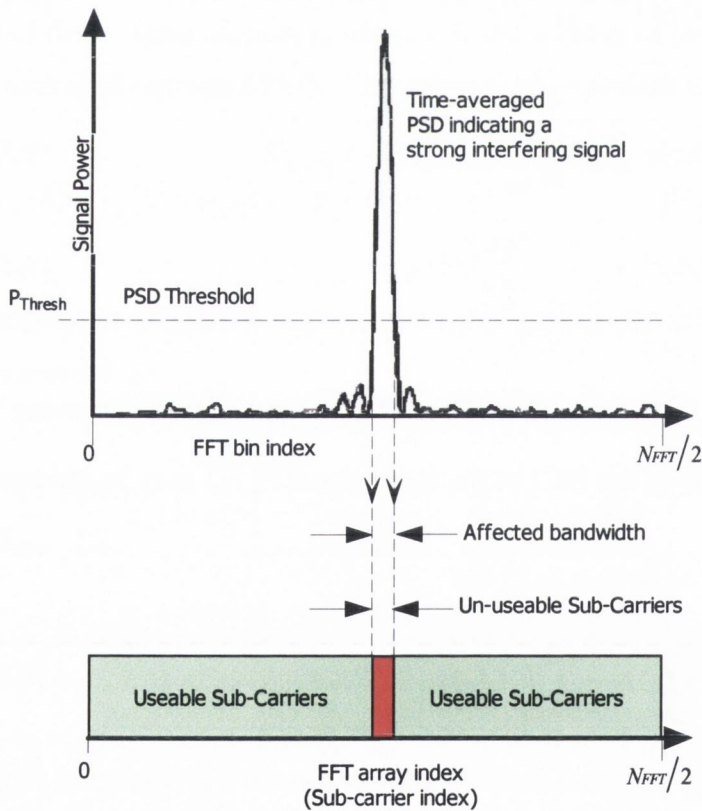


Figure 9.3: Sub-carrier allocation using the time-averaged PSD information.

The proposed sub-carrier allocation scheme is essentially an ‘interference-avoidance’, or ‘carrier-avoidance’ scheme as ‘busy’ or excessively noisy frequencies are avoided. It is

important therefore, that there are no transmissions from transceivers associated with the intended reconfigurable radio application during the channel-monitoring process. This is required in order to ensure that only frequencies experiencing unrelated interference and noise are avoided instead of valid sub-carriers in use by other associated transceivers.

An example of how the time-averaged PSD information may be used to allocate sub-carrier is shown in Figure 9.3. This example illustrates that a strong interfering signal affecting a specific frequency range. If a reconfigurable OFDM transceiver attempted to transmit information using sub-carriers in this range, the possibility that this transmitted information would not be correctly received at the destination or un-recoverably corrupted by the strong interference is high. In order to avoid this frequency range, the sub-carrier allocation scheme deems the sub-carriers with a PSD greater than the threshold value P_{thresh} , as ‘unusable’. These unusable sub-carriers are shown in red in this diagram and the useable sub-carriers are shown in green.

Sub-carrier allocation occurs on a per-OFDM frame basis in the reconfigurable OFDM transceiver presented in this thesis. The time-averaged PSD of the received signal, represented by the array $\hat{P}(f_k)$ is therefore used to obtain the channel mask before each new OFDM frame is transmitted. A PSD threshold value, determined by experimentation is used to decide whether a sub-carrier may, or may not be, allocated for use in the reconfigurable OFDM transceiver. If the estimated PSD of a sub-carrier exceeds this predefined threshold value, then the sub-carrier is deemed ‘unusable’ by storing a binary 0 in the channel mask array index corresponding to the sub-carrier index. Conversely, a sub-carrier may be deemed ‘useable’ if the estimated PSD corresponding to the sub-carrier is below the predefined PSD threshold value.

9.5.1 Reconfiguration

When all of the $N_{FFT} / 2$ possible sub-carriers have been analysed and a decision regarding their viability for information transmission has been made, the information sources requiring transmission must then be mapped to the allocated (useable) sub-carriers. Transceiver reconfiguration is therefore required in the form of:

- Reconfiguration of the Serial to Parallel (S/P) conversion stage.

- Reconfiguration of the IFFT stage.

9.5.1.1 Reconfiguration of the Serial to Parallel (S/P) conversion stage

This stage is illustrated in Figure 9.1, and as discussed in Chapter 2, performs the conversion of the modulated input sequence into a number of parallel subsets of the modulated input sequence. In this case, the number of parallel modulated signal subsets is dependent on the number of sub-carriers available for transmission. In other words, if N_{SC} denotes the number of useable sub-carriers obtained before each OFDM frame transmission by the proposed sub-carrier allocation scheme, then the modulated input signal sequence may be converted to N_{SC} parallel sub-sets of a set of the input sequence. As the value of N_{SC} may vary due to the time-varying characteristics of the monitored frequency band, the corresponding number of parallel sub-sets may also vary. This means that the Serial to Parallel stage shown in Figure 9.1 may need to be reconfigured before the transmission of a new OFDM frame in order to accommodate the time-varying value of N_{SC} .

9.5.1.2 Reconfiguration of the IFFT stage

The ‘useable’ sub-carriers chosen from the total set of $N_{FFT} / 2$ possible sub-carriers are represented by the array of binary values comprising the channel mask. If the n^{th} binary value contained in the channel mask array is 0, then the sub-carrier represented by the n^{th} index of the FFT array will not be used. If the n^{th} binary value contained in the channel mask array is 1, then the sub-carrier represented by the n^{th} index of the FFT array will be used. This operation may be achieved by logically *AND*-ing the FFT array representing the $N_{FFT} / 2$ sub-carriers with the channel mask. The FFT array may be viewed as an array of $N_{FFT} / 2$ binary ones as all the sub-carriers may be potentially used. Therefore the logical *AND* operation produces the channel mask. The direct channel mask to sub-carrier mapping ensures that the data symbols from each of the N_{SC} parallel modulated sub-sets of the original set of the input signal are mapped to the FFT array index. Subsequently, this means that the correct sub-carriers will be used to transmit the information.

The IFFT stage may be reconfigured if the allocated sub-carriers are changed following the transmission of the previous OFDM frame. Reconfiguration comprises activating or deactivating sub-carriers using the channel mask information. If a greater number of sub-carriers are required for transmission, then the size of the IFFT array must be modified. This involves allocating a new IFFT array of a size greater (or smaller) than the existing IFFT array in order to accommodate the change of sub-carrier numbers. As a result of this, the FFT size must be changed to accommodate the increased/decreased number of sub-carriers, therefore the length of the output sequence of the IFFT stage also changes.

9.5.2 Receiver Notification and Synchronisation

The receiver must be able to correctly extract, reassemble and demodulate the received information represented by the sub-carrier values. The sub-carriers, which are used to convey information from the source to one or more remote reconfigurable OFDM transceivers may change each time a new OFDM frame is transmitted. A consequence of this is that the sub-carrier allocation information must be passed to the remote recipients. The remote recipients may then use this information to correctly extract the information conveyed by the sub-carriers that were allocated at the source transceiver.

The proposed sub-carrier allocation technique operates on a per-OFDM frame basis. This means that the sub-carrier allocation information must be conveyed to the remote transceivers prior to the transmission of a new OFDM frame by the source reconfigurable OFDM transceiver. As part of the frame synchronisation and DOFDM techniques presented in Chapter 7 and Chapter 8, respectively, a frame synchronisation training symbol is transmitted before each new OFDM frame. As discussed previously, this high-power training symbol enables the receiver to firstly detect the training symbol and secondly, estimate where the start of the OFDM frame occurs in the intercepted signal sequence. One of the main aims of the proposed sub-carrier allocation scheme is that it should not contribute to the transmission overheads of the reconfigurable OFDM transceiver. This means that the amount of information that may then be transmitted may be maximised.

The proposed receiver notification and synchronisation method increases the usefulness of this frame synchronisation symbol. The proposed idea combines frame synchronisation, carrier frequency offset estimation and sub-carrier allocation notification using a single

OFDM training symbol. The transmission overheads may be significantly reduced as only one single OFDM training symbol is required to accomplish three important tasks.

These three tasks are:

- Frame synchronisation.
- Carrier-frequency offset correction.
- Receiver notification of the currently allocated sub-carriers.

Upon notification of which N_{SC} sub-carriers are being currently used to transmit information in the next OFDM frame, the receivers may then reconfigure themselves using this information in order to correctly extract the information contained in these N_{SC} sub-carriers.

9.5.3 Enhanced Training Symbol (ETS)

The term Enhanced Training Symbol (ETS) will be used to refer to the frame synchronisation training symbol incorporating the sub-carrier allocation information. In order to select the correct N_{SC} sub-carriers from the total set of $N_{FFT}/2$ sub-carriers, each of the remote transceivers using this system must obtain the channel mask. The channel mask, shown in Figure 9.2, consists of $N_{FFT}/2$ binary values. This enables the remote transceivers to select the correct sub-carriers for demodulation during the FFT stage in the receiver (assuming that all of the associated transceivers are using the same FFT size).

The $N_{FFT}/2$ binary values contained in the channel mask array may be converted to a set of pilot symbols. For example, if $N_{FFT} = 32$, then the number of total sub-carriers that may be used for transmission is $N_{FFT}/2 = 16$. The channel mask associated with these carriers therefore contains 16 binary values, with one binary value representing each of the sub-carriers. These 16 binary values may be represented in modulated forms as 8 QPSK symbols, or 4 16-QAM pilot symbols, where one QPSK symbol represents two bits and a 16-QAM symbol represents four bits. The frame synchronisation training symbol, as described in Section 7.3, is comprised of two identical symbol halves. To briefly recap:

Each of the identical frame synchronisation symbol halves are composed of a number of pseudo-randomly generated pilot symbols taken from a high-order constellation diagram. This constellation diagram corresponds to a modulation scheme with a high bit-to-symbol

ratio. The chosen pilot symbols correspond to high-energy or high-power signals from the constellation diagram of choice in order to ensure that the receiver detects the training symbol.

Instead of using pseudo-randomly generated pilot symbols for the frame synchronisation technique, the modulated channel mask values are used as the pilot symbols. In order to maintain the high-power OFDM training symbol requirement, the channel mask binary values are modulated using 16-QAM. This results in 4 pilot symbols. The frame synchronisation algorithm described in Chapter 7, constructs an ETS comprising of two identical half symbols. This is illustrated in Figure 9.4, which shows the identical ETS halves with the pilot symbols illustrated in green. In this example, the number of sub-carriers that may be used is $N_{FFT}/2 = 16$. The 4 pilot symbols are then inserted into the FFT array for each identical half of the ETS.

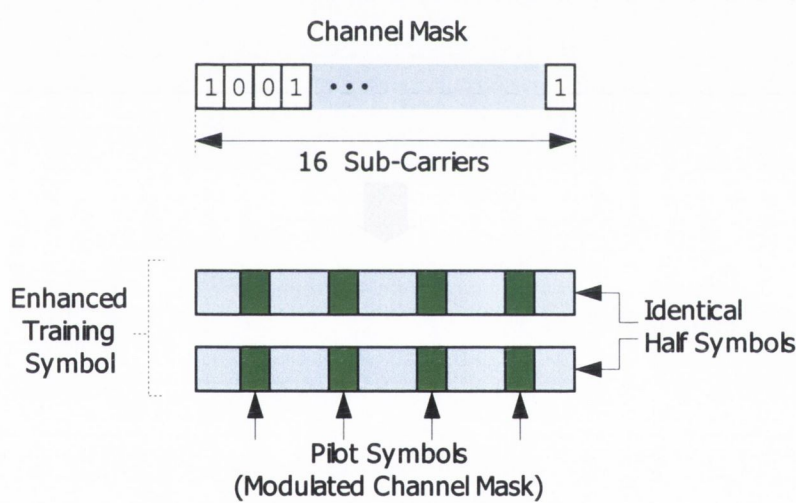


Figure 9.4: Construction of the ETS from the channel mask.

Upon reception of an ETS, the receiver may therefore estimate the start of the OFDM frame as previously described in Chapter 7. In addition to frame synchronisation, carrier frequency offset compensation may also be carried out. The channel mask values may be obtained by demodulating the received ETS. It is important therefore that the structure and modulation scheme used in the creation of the ETS are known to all of the remote transceivers.

9.6 Implementation

A block diagram of how this sub-carrier allocation technique may be implemented using the IRIS platform to form an enhanced reconfigurable OFDM transceiver is shown in Figure 9.5. This is the base-band model of the complete transceiver and it is implemented using a 2 GHz Pentium IV GPP.

Only the section of the reconfigurable radio that exists purely in the software domain is illustrated in this diagram (i.e. the baseband signal-processing functions and related Control Logic functions). The ADC/DAC and RF front-end are not considered in this case. This diagram is developed from Figure 9.1 to include the frame synchronisation and dynamic sub-carrier allocation stages. The signal-processing stages shown in the yellow boxes are those that are directly relevant to the sub-carrier allocation technique presented in this section. The stages illustrated in green denote the Control Logic functions that are essential for receiver and transmitter coordination. The Radio Components shown in the blue boxes are other essential components of this implementation but are not directly relevant to the sub-carrier allocation scheme being focussed on in this section.

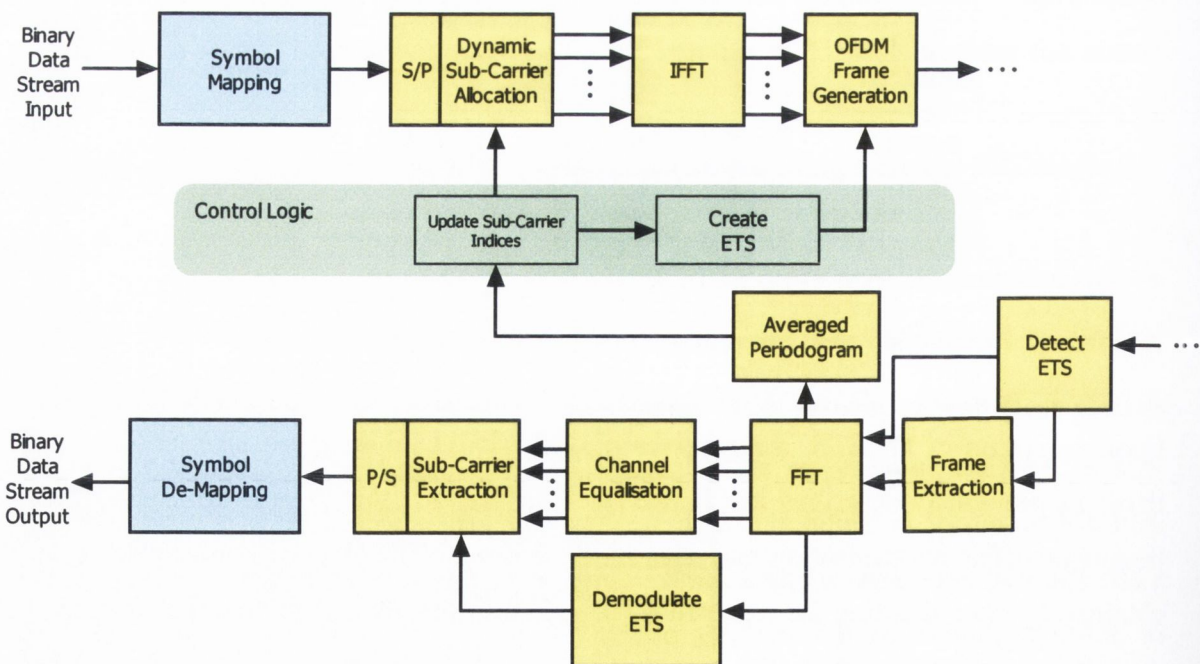


Figure 9.5: Sub-carrier allocation and ETS creation/demodulation using IRIS.

Beginning with the OFDM modulator, a sequence of binary information from the source is converted to a number of shorter binary values sub-sets in parallel. This is achieved using the serial to parallel conversion stage (S/P). The Averaged Periodogram stage estimates the time-averaged PSD of the frequency band of interest. This information is then used by the Update Sub-Carrier Indices Control Logic function to reconfigure the Dynamic Sub-Carrier Allocation stage in the transmitter. Reconfiguration involves establishing the data symbol to sub-carrier mapping using the channel mask. The IFFT stage creates the sequence of OFDM symbols in parallel and the OFDM Frame Generation stage converts these parallel OFDM symbols to a serial sequence of OFDM symbols. The final stage in the transmitter section illustrated in this diagram is to insert the ETS. This ETS functions as a frame synchronisation, carrier frequency offset estimator and sub-carrier notification tool. The ETS may be demodulated to obtain the channel mask denoting the allocated sub-carriers for this particular OFDM frame. This ETS is inserted in the frame guard prior to the OFDM frame.

Upon reception of the ETS in the intercepted, down-converted and digitised signal, the Frame Guard Extraction stage extracts the sequence of OFDM symbols from the OFDM frame. This series of OFDM symbols is then passed to the FFT stage. Considering one OFDM symbol at a time, the FFT stage de-multiplexes the sub-carriers contained in the OFDM symbol. The ETS is also de-multiplexed using the FFT stage. The pilot symbols contained in the ETS are then extracted and demodulated to obtain the channel mask. Channel equalisation and carrier frequency offset estimation and compensation may be performed using the Channel Equalisation stage. When the channel mask has been obtained, the information that was transmitted using the allocated sub-carriers may then be extracted from the FFT array (using the Sub-Carrier Extraction stage). These complex-valued data symbols in parallel are then converted to a serial sequence. The final stage, in regards to information retrieval is to demodulate the data symbols using the Symbol De-Mapping stage. Following synchronisation, the sub-carrier allocation information may be updated during the null signal portion of the frame guard interval (i.e. no transmissions other than noise and unrelated interference are present on the communications channel).

9.7 Evaluation

One objective of this sub-carrier scheme is to choose frequencies that are not subject to significant amounts of interference or noise. This means that the power required to

transmit information to the remote receivers may be reduced while still potentially achieving a desired BER at the receiver. This is important as the sub-carrier allocation process is being carried out by only one transceiver in this scenario. The time-averaged PSD of the channel, which the proposed sub-carrier allocation scheme is based on, is specific to the wireless channel that is 'seen' by the source transceiver. This scheme is designed for short-range high data rate wireless communications. Therefore, it is expected that there is a strong correlation between the spectral activity in the form of noise and unrelated interference experienced by source and remote transceivers. In other words, the source of the interference or noise may be considered to be common to all of the transceivers using this sub-carrier allocation scheme. Therefore, if one or more sub-carriers are deemed unusable by the source transceiver, then there may be a high possibility that the destination (remote) transceivers also deem these sub-carriers as being unusable.

Reconfiguration means that one or all of the Radio Component functions illustrated in Figure 9.5 may be modified, replaced, or even the structure of this baseband transceiver model may be dynamically changed. It is possible is even replace one or more of the Radio Components if a less complex or more robust signal-processing stage is required. In terms of the sub-carrier allocation technique presented in this section, this means that the algorithm may be modified to increase the performance and processing power efficiency of the scheme. By lowering the PSD threshold, P_{thresh} , the sensitivity of the sub-carrier allocation scheme to lower-level channel activity may be raised. This may be used to ensure that only sub-carriers with a very low PSD over the duration of the periodogram averaging filter are allocated for use for a noise/interference intolerant application. Conversely, P_{thresh} may be increased for more noise/interference tolerant applications are required.

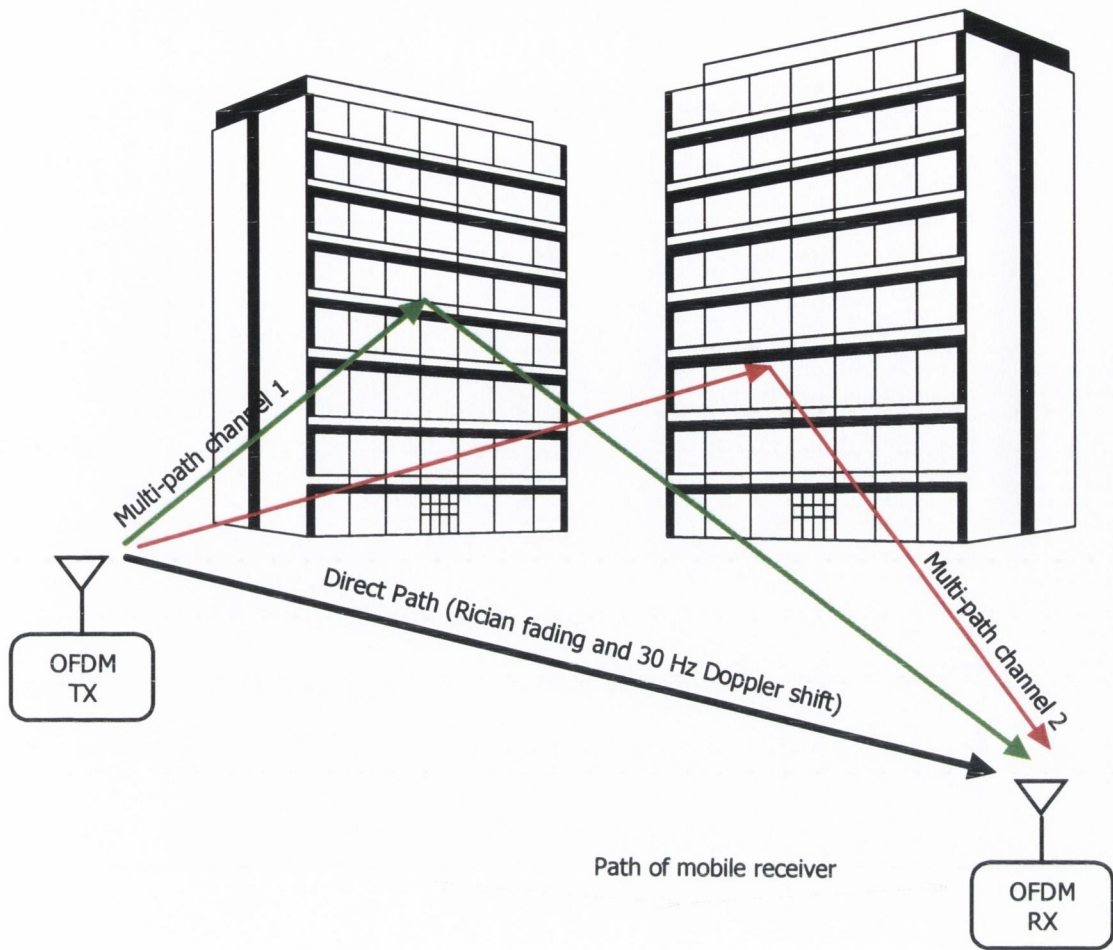


Figure 9.6: Model of mobile receiver experiencing direct path and two multi-path received signal components from a source OFDM transmitter.

Figure 9.6 illustrates the propagation model used for simulations of this sub-carrier allocation scheme. The results presented in this section focus on the BER of a series of received channel masks obtained by demodulating the ETS. It is assumed that perfect frame synchronisation and carrier frequency offset has been achieved and that the only sources of errors in the received channel mask values are due to the effects of the channel. This channel is shown in block diagram form in Figure 9.7.

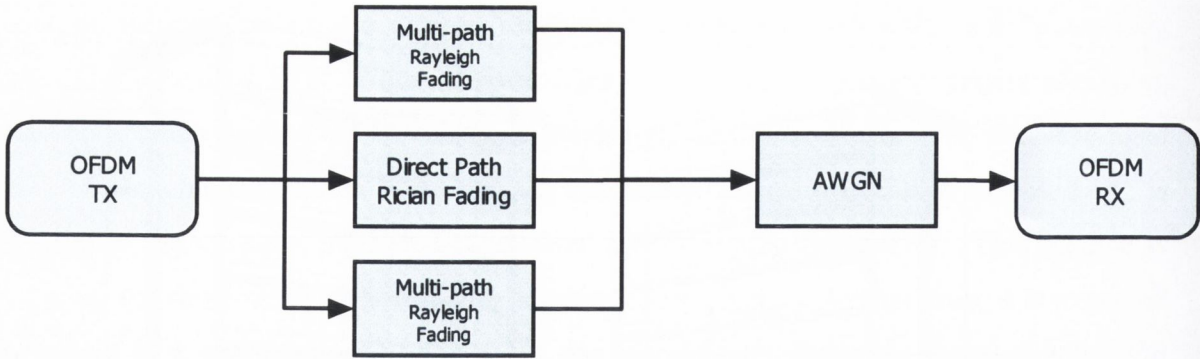


Figure 9.7: Block diagram of combined multi-path, direct path, Doppler fading and AWGN channel model used for sub-carrier allocation simulations.

The channel model as shown in this diagram consists of a direct-path signal that is subjected to Rician fading and a Doppler shift of 5 Hz, two multi-path Rayleigh fading signal components, which are also subjected to Doppler shift and an AWGN channel. These channel conditions are summarised in Table 9.1. The simulation tests involve varying the SNR of the AWGN channel model from 40 dB to 0 dB and measuring the BER of 100 received channel masks.

Table 9.1: Table of multi-path and direct path channel conditions used for simulations.

Propagation Mode	Fading	Gain (dB)	Delay (ms)	Doppler (Hz)
Direct Path	Rician	0	0	5
Multi-path	Rayleigh	-10	2	5
Multi-path	Rayleigh	-20	4	5

The channel mask modulation scheme used for the simulation tests is non-coherent QPSK and the channel mask values being transmitted are not encoded. Figure 9.8 is a graph of the BER of the received batch of channel masks versus the SNR. The initial SNR value of 70 dB was used in the simulations to ascertain the BER for an almost noise-less case. This figure shows that the fading direct path, two multi-path components and combined Doppler fading are the main reason for the BER of approximately 0.05 for the 70 dB case. More realistic simulations involve the BER for the 40 dB to 0 dB scenarios. The BER may be deemed excessive when the SNR is reduced to approximately 20 dB. At this stage, the BER is approximately 0.2. It is concluded that a high SNR value is required to ensure that un-encoded, non-coherent transmission of QPSK modulated channel mask values are

received with a minimal acceptable error by the remote transceivers employing this sub-carrier allocation scheme.

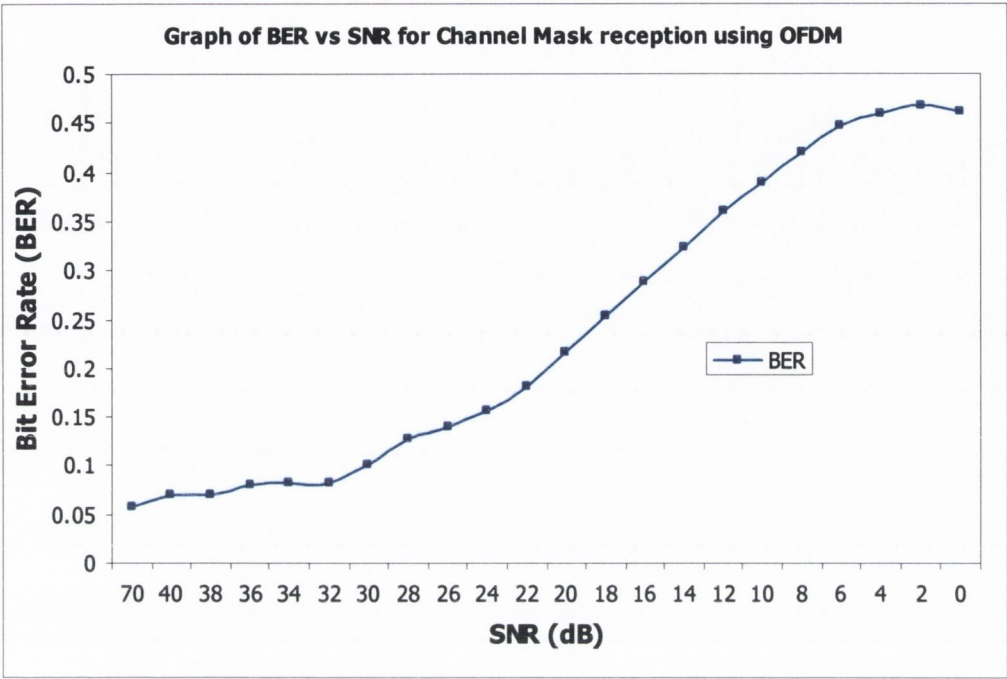


Figure 9.8: Graph of the BER vs. SNR for 100 QPSK modulated channel masks over an AWGN channel with multi-path Rayleigh fading, direct-path Rician fading and Doppler fading effects.

Of direct relevance to this thesis is a performance comparison of the dynamically-allocated OFDM scheme versus a normal static sub-carrier allocation OFDM. The test scenario in this case is an AWGN channel with strong FM interference in the intended OFDM transmission band. Two of the main areas of interest are in relation to the improvement in BER and increase in data-throughput when using the dynamically-allocated OFDM scheme compared to the normal static sub-carrier allocation OFDM case. The dynamically-allocated OFDM scheme avoids sub-carriers with a PSD exceeding a specific threshold value while the normal static-OFDM case uses all specified sub-carriers regardless of the current spectral activity on the band of interest.

The AWGN channel model under consideration in this case is affected by an interfering FM transmission comprising three tones of equal power. This interference source has a peak power of 2.9dBm and the centre frequency is located within the desired OFDM transmission band. The first test scenario involves measuring the BER and data-throughput of OFDM using fixed sub-carriers that do not adapt to the current wireless

channel activity and compete with the interference source. This normal statically-allocated OFDM transmission uses 48 sub-carriers, where each sub-carrier is modulated using QPSK. The peak power of this OFDM signal is 0.16dBm. The second scenario involves the same interference-affected AWGN channel model but the OFDM signal is changed. In this case, the dynamically allocated OFDM sub-carriers, chosen from a maximum possible set of 48 carriers, avoid the frequencies experiencing interference with a PSD greater than approximately 1% of the maximum estimated peak power of the interfering signal. The implication of the interference-avoidance tactic is that potentially less sub-carriers will be used but the BER is expected to be reduced due to the interference-avoidance properties of this scheme. The subcarriers used for this scenario are therefore modulated using 16-QAM, a higher order modulation scheme than that used for the fixed sub-carrier OFDM scenario in order to increase the potential data-throughput that can be supported by the expected BER reduction. The peak power of the dynamically-allocated OFDM scheme is 1.4dBm.

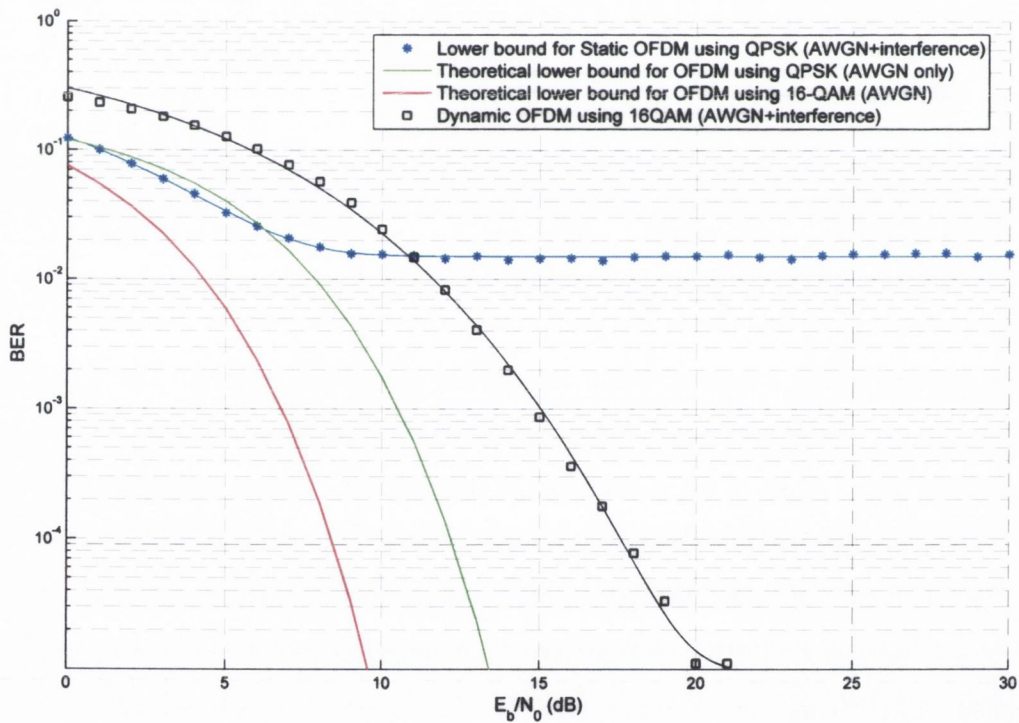


Figure 9.9: Graph of the BER vs. E_b/N_0 showing the lower bounds for OFDM employing QPSK and 16-QAM over an AWGN channel. The lower bound for normal OFDM (static sub-carrier allocation) over an AWGN channel affected by strong interference from an FM transmission is shown in blue. The lower bound of the dynamically-allocated OFDM scenario employing 16-QAM is shown in black.

Figure 9.9 is a graph of the BER versus E_b/N_0 illustrating the performance of the statically-allocated OFDM scenario where the sub-carriers are modulated using QPSK

compared to the performance of the dynamically-allocated interference-avoidance OFDM scheme using 16-QAM modulated sub-carriers. For each E_b/N_0 increment in this simulation, 1000 OFDM frames are transmitted and received and perfect synchronisation is assumed. The theoretical lower bounds for QPSK and 16-QAM over an interference-free AWGN channel for $0 \leq E_b/N_0 \leq 30$ are shown by the green and red graphs, respectively. The effect of the interference is to increase the BER for the statically-allocated OFDM case to approximately 0.015 for $E_b/N_0 \geq 8$. This is illustrated by the blue graph in Figure 9.9. The dynamically-allocated interference-avoidance OFDM signal is not significantly affected by the interference and the BER is approximately 5.5×10^{-6} for $E_b/N_0 \approx 20$.

One other important outcome from these tests is the difference in data-throughput between the statically-allocated OFDM case and the dynamically-allocated interference-avoidance OFDM technique. Regarding the former, the sub-carrier modulation scheme used is QPSK and for $E_b/N_0 \approx 20$, the BER is approximately 0.015 and the total uncorrected data-throughput is 96000 bits. Regarding the dynamically-allocated interference-avoidance OFDM case, the sub-carrier modulation scheme used is 16-QAM. For $E_b/N_0 \approx 20$, the BER is approximately 5.5×10^{-6} and the total number of uncorrected received bits is 180000.

9.8 Conclusions

This chapter has presented a sub-carrier allocation scheme that operates on a per OFDM-frame basis and may be used to improve the robustness of OFDM in noisy and interference-prone frequency bands. This is achieved by enabling the source OFDM transceiver to avoid sub-carriers with a time-averaged PSD that exceeds a threshold value. Each of the destination OFDM transceivers within range of the source transceiver may obtain the sub-carrier allocation information used for an OFDM frame by demodulating the ETS. As explained in the previous chapter, the ETS is also used for frame synchronisation and carrier-frequency offset estimation. In effect, a single ETS may perform three key tasks during the OFDM demodulation process.

It has been shown that the enhanced OFDM technique using dynamically-allocated sub-carriers for interference-avoidance purposes results in a significantly reduced BER and significantly increased data-throughput over an AWGN channel affected by multi-

frequency interference. This technique also presents a solution to the problem of achieving interference-free transmission mode sharing on a common RF band.

This sub-carrier allocation scheme may be developed further by enabling the amount of information transmitted on each of the sub-carriers to vary. By being able to classify a frequency band in terms of individual sub-carrier quality, it is possible to employ different modulation schemes for each sub-carrier. For sub-carriers with a low time-averaged PSD, a complex modulation scheme with a high bit-to-symbol ratio may be used. An example of this modulation scheme is 64-QAM, which enables 8 binary values to be represented as one data symbol. A sub-carrier with a very low time-averaged PSD is understood to mean a high-quality sub-carrier in this thesis. These high-quality sub-carriers may be able to support a modulation scheme with a high bit-to symbol ratio such as 64-QAM, as the estimated noise and/or interference level may not sufficient enough to have an adverse affect on the BER of the received information. Received information using a sub-carrier with a higher time-average PSD may experience a high BER if a modulation scheme with a high bit-to-symbol ratio is used. Therefore, in order to increase the robustness of the transmission system and achieve a desired BER using a noisy sub-carrier, a modulation scheme with a low bit-to-symbol ratio may be used. An example of this is BPSK (one bit per data symbol).

In addition to the sub-carrier allocation information, the remote transceivers also require knowledge of the modulation schemes associated with each of the sub-carriers. Another main objective is to ensure that the transmission overheads required to facilitate this feature are minimised. The following section presents a technique that may be used by the remote transceivers to automatically recognise the modulation scheme used for each of the sub-carriers. This proposed scheme means that the source transceiver may not have to transmit extra information regarding the modulation schemes used for each of the sub-carriers.

10

MODULATION SCHEME RECOGNITION

10.1 Introduction

This chapter introduces the idea of automatic modulation scheme recognition. Automatic modulation scheme recognition enables a receiver to estimate what digital modulation scheme has been used in an unknown intercepted signal. This allows the receiver to demodulate a received signal without *a priori* knowledge of how the signal was modulated at the source. The transmitter may then be reconfigured using this information to transmit signals that match the modulation scheme used in the intercepted signal.

The work presented in this chapter has been published in Proceedings of the First Joint Institute of Engineers of Ireland IEI/IEE Symposium on Telecommunications Systems Research [Nolan2001], Proceedings of the Irish Signals and Systems Conference (ISSC) in 2002 [Nolan2002a], Proceedings of the IASTED International Conference on Signal Processing, Pattern Recognition, and Applications (SPPRA 2002) [Nolan2002b], and Proceedings of the IEEE Wireless Communications and Networking Conference WCNC 2002 [Nolan2002c].

10.2 Placing the Work in Context

It was concluded in the previous chapter that different modulation schemes may be used to transmit information on the allocated sub-carriers. This means that the overall transmitted data throughput rate may be increased as high quality sub-carriers may be capable of supporting modulation schemes with a high bit-to-symbol ratio. Automatic modulation scheme classification enables the receiver to estimate the type of modulation used for each

10.3 Review of Modulation Recognition Schemes

The area of modulation scheme recognition has generated a significant amount of research work and proposals over the past number of years. Processing power and computational capabilities have dramatically increased over the past ten years meaning that modulation scheme classification techniques have been moving from the purely theoretical domain to real-time implementations. It is now possible to incorporate a greater number of real-time modulation scheme recognition techniques into a reconfigurable radio. The result of this is that a radio that is significantly more adaptable than traditional hardware-only radio systems may be created.

The vast majority of modulation scheme classification techniques may be classed as decision-theoretic, statistical moments, spectral or pattern recognition-based approaches. Decision-theoretic-based methods rely on examining the statistical properties of the waveform. These properties may include the phase and amplitude of the signal. Pattern recognition-based classifiers rely on ‘features’ of the intercepted waveform to create a waveform signature. The pattern recognition algorithm then attempts to match the signature of the unknown waveform to an inventory of known waveforms.

One approach taken to developing a modulation scheme classification technique is pattern recognition [Jondral1989][Liedtke1984]. Jondral’s [Jondral1989] proposed technique was designed to classify 2-state Amplitude Shift Keying (ASK2), BPSK, Frequency Shift Keying (FSK2) and FSK4 modulation scheme types in addition to Amplitude Modulation (AM) and Single Sideband (SSB) analogue modulation schemes. Key features of the received signal; instantaneous amplitude, frequency and phase are used to discriminate between the different modulation scheme types. A later proposal by Ketterer, Jondral *et al* [Ketterer1999] uses a time and frequency distribution called the Cross Margeneau-Hill distribution to extract the amplitude, phase and frequency information in a received signal. Of interest is the reported very low Error Classification Rate (ECR) for simulated BPSK, QPSK, 8-PSK and 16-PSK signals. Details of the simulated channel environment are not given. Simulated shortwave signals are used for part of the testing although it is not clear whether the challenging multi-path, interference-prone and noisy wireless channel environment associated with signals below 30 MHz is considered in the simulation results.

Lopatka and Pedzisz [Lopatka2000] proposed a two-step classification method for 4-state Differential Phase Shift Keying (4DSPK), 16-QAM and FSK. This scheme obtains the statistical moments of amplitude, phase and frequency of the unknown signal before passing this information to the pattern recognition stage. Lopatka reports that only *a priori* knowledge of the carrier frequency is required. Experimental results presented in this paper, considering an AWGN channel only state that the lower-bound SNR for correct identification is 5 dB.

Another modulation classification technique, based on Hellinger distance [Beran1977] [Shiryaev1995] parameter estimation, is reported to be able to automatically overcome a moderate degree of noise model distortion, with improved robustness and efficiency [Anh1997] [Donoho1997].

Kim and Polydoros [Kim1988] compare three different types of modulation scheme classifiers and derive a quasi log-likelihood (qLLR) functional for BPSK and QPSK classification. The three classifiers compared in Kim's paper are decision-theoretic, square-law and phase-based metrics. Classification algorithms taking a decision-theoretic rely on calculating the likelihood ratio of an unknown waveform being modulated using a particular modulation scheme. For BPSK and QPSK schemes, square-law modulation scheme estimators base their estimates on the spectral content of the square of the signal. In the absence of noise, a BPSK signal will exhibit an increase in the spectral energy of the signal (a harmonic) at twice the carrier frequency whereas QPSK will not. Reichert [Reichert1992] proposed a statistical moment-based classifier in 1992 using this square-law idea. A phase-based classifier exploits the fact that the number of clusters of signal points on a constellation diagram is a function of the modulation scheme (e.g. a BPSK signal exhibits two such signal point clusters and QPSK has four). This approach is of particular relevance to the modulation classification technique presented in this section. Kim and Polydoros's proposed technique relies on the assumption that the carrier frequency, Carrier to Noise Ratio (CNR), symbol rate and initial phase are known by the receiver. This information may not be available when considering a reconfigurable radio implementation in an unknown channel environment.

High-accuracy modulation scheme classification techniques have also been based on Artificial Neural Networks (ANN) [Haykin1999]. Ghani and Lamontagne [Ghani1993] propose an ANN-based classifier for FSK, BPSK and QPSK schemes but note that

extensive modulation classification training has to be undergone before the technique may be applied for real-signals. They also note that a significant level of noise reduction and enhancement of the features used to discriminate between modulation schemes is required. A similar ANN-based scheme proposed by [Arulampalam1999] describes the signal features that may be used to aid the signal classification. Signal classification may be based on the standard deviation of the phase, frequency and amplitude of the non-weak segments of the intercepted signal. In addition, the maximum PSD value of the normalised signal may also be used. Ta [Ta1994] uses wavelet analysis to classify PSK, FSK and ASK signals. This wavelet packet deconstruction method using an ANN but without pre-processing was tested using an AWGN channel with a minimum SNR of 10 dB. Requiring extensive waveform training, this scheme is reported to achieve a modulation scheme estimation accuracy of over 90%. The pre-processing required for most of the ANN-based estimators and training processes are processor-intensive tasks. As a result, these techniques are best suited for off-line modulation scheme classification and not for use in the reconfigurable OFDM radio presented in this thesis.

Real-time modulation scheme recognition techniques have been proposed by Boudreau, Dubuc, *et al* [Boudreau2000] using dual processors for a spectrum-monitoring application. A modulation scheme recognition technique based on the zero-crossings of an intercepted signal was proposed by Hsue and Soliman [Hsue1990] in 1990. and also by Hsue, Soliman [16] using three processors and parallel processing approaches.

Signal space representation of quadrature components may be referred to as the constellation diagram of a particular modulation scheme. This is a graphical means of examining the phase and amplitude information of a digitally modulated signal. Constellation diagrams are commonly used to assess the underlying signal structure of a signal. The number of expected signal point clusters increases linearly as complexity of the M -ary PSK/ M -ary QAM scheme increases. Using this expected signal point clusters, a statistical moment-based approach may then be taken in order to estimate the modulation scheme associated with the unknown signal.

10.4 Proposed Modulation Scheme Classification Technique

The modulation scheme recognition technique presented in this thesis uses the phase and amplitude information of a normalised intercepted digitally-modulated intercepted signal.

A constellation diagram may be used to represent the phase and amplitude the incoming signal. This is referred to as a 'signal-space' representation of the received signal. The signals of interest are digitally modulated waveforms where each modulated symbol is associated with a discrete phase and/or amplitude distribution. Examples of these modulated waveforms include BPSK, QPSK, 8-PSK, 16-QAM, 64-QAM and other M -ary PSK/ M -ary QAM schemes. The proposed classification technique calculates the moments of the normalised signal points about a specific point on the constellation diagram. The moment values are expected to be part of a monotonically increasing sequence as M increases. This proposed technique is also designed to present a significantly lower processing-time overhead to the reconfigurable radio system. Most importantly, the estimation time must be minimised in order to attempt to demodulate multiple parallel signal sequences that occur when dealing with adaptively modulated sub-carriers in a reconfigurable OFDM system. The combined waveform type estimation and demodulation processes must be completed before the next OFDM symbol or OFDM frame is received. In an attempt to achieve this reduced waveform estimator complexity, the set of possible signal points is reduced. This means that for an M -ary modulation scheme (e.g. BPSK, where $M = 1$), the number of possible states on a constellation diagram is reduced to $M/2$ states. The moment of the remaining signals points is reduced by two therefore the number of calculations required for an estimated may also be reduced. This modified constellation diagram is referred to as a Reduced Form Constellation Diagram (RFCD) in this thesis.

10.4.1 Reduced Form Constellation Diagram (RFCD)

Consider the case where all of the possible transmitted symbols from an M -ary PSK alphabet are equi-probable and are affected by AWGN. Let c_I and c_q denote the In-Phase (I) and Quadrature (Q) components of one of the received pilot/data symbols following the FFT stage of the reconfigurable OFDM receiver. The time-varying sequence of data/pilot symbols are normalised to unity scale in order to maintain modulation scheme estimation consistency and accuracy. This means that the set of N received data/pilot symbols denoted by R , where:

$$R[0, \dots, N-1] = c_I[0, \dots, N-1] - jc_q[0, \dots, N-1] \quad [10.1]$$

is replaced by R_{scaled} , where:

$$R_{scaled}[0, \dots, N-1] = \frac{K |c_i[0, \dots, N-1] - jc_q[0, \dots, N-1]|}{\text{Max} \left\{ \left(c_i[0, \dots, N-1] - jc_q[0, \dots, N-1] \right)^2 \right\}} \quad [10.2]$$

and where K is the scaling factor.

Since all of the possible received data/pilot symbols are considered equi-probable, then the expected signal space, or signal constellation representation of these data/pilot symbols is expected to be approximately symmetrical.

The RFCD may therefore be obtained by discarding signal points that do not satisfy the following condition: $\Re\{R_{scaled}[0, \dots, N-1] \geq 0\}$, where $\Re[\cdot]$ denotes the real part of the entity enclosed in square brackets.

The set of original N data/pilot symbols is therefore reduced to the normalised RFCD set of symbols denoted by:

$$R_{RFCD}[0, \dots, L-1] = \Re\{R_{scaled}[0, \dots, N-1] \geq 0\} \quad [10.3]$$

where L is the number of real-valued symbol samples contained in the array given by R_{scaled} .

An example of the RFCD approach applied to a signal sequence modulation using QPSK and affected by AWGN is shown in Figure 10.2. The signal values with a real part that is less than zero are disregarded.

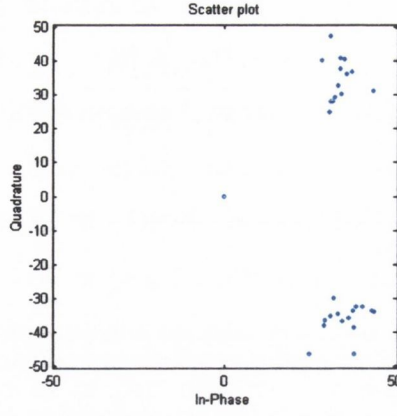


Figure 10.2: Reduced Form Constellation Diagram of a QPSK signal sequence affected by AWGN.

10.4.2 Statistical Moment Calculations

The p^{th} order statistical moment I_{0xy} is the sum of the moments about the point $[x = \text{Max}\{\Re(R_{RFCD})\}, y = 0]$ of the signal space diagram representation of the RFCD set of symbols.

I_{0xy} is defined as:

$$I_{0xy} = I_{0x} + I_{0y} = 2 \sum m_i r_i^p \quad [10.4]$$

where the ‘weighting’ of each of the signal points, $m_i = 1$ for all of the signal points in the RFCD set and

$$r_i = |(R_{RFCD}(i))^2|, \quad 0 \leq i < L \quad [10.5]$$

The classification technique is threshold based, where the threshold value, M_{thresh} is obtained through testing. For one scenario where the input signal may be modulated using BPSK or QPSK, the expected statistical moments of the QPSK signal points about the point $0.5 + 0j$ will be greater than the expected statistical moments of the BPSK signal points. The threshold value M_{thresh} may then be designated as the statistical moment that is greater than the largest statistical moment value associated with the BPSK scheme. Therefore, if the calculated statistical moments of the intercepted signal are greater than M_{thresh} , the modulation scheme is classed as a QPSK scheme. Conversely, if the

calculated moment value is less than M_{thresh} , then the signal may be classed as a BPSK waveform.

10.5 Implementation

Figure 10.3 illustrates how the proposed modulation scheme recognition technique is incorporated into the reconfigurable OFDM transceiver using the IRIS framework on a GPP platform. In this example, only the relevant sections of the receiver are shown. Following the Sub-Carrier Extraction stage, an extra Radio Component is inserted. This Radio Component, implemented using the generic IRIS DSP component framework, encapsulates the proposed modulation scheme classification technique within the associated `Process()` function call.

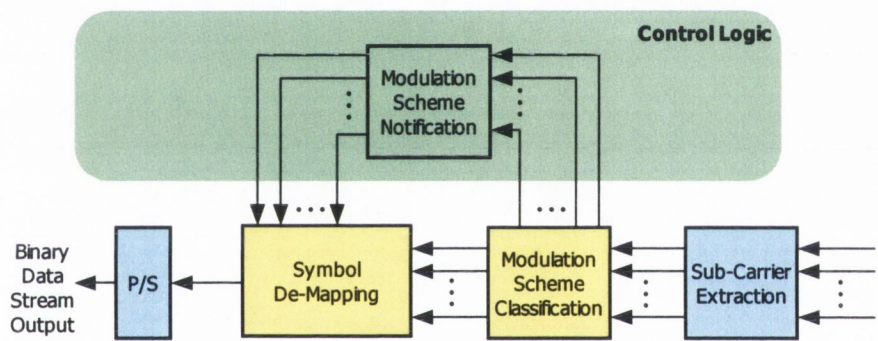


Figure 10.3: Block diagram of the reconfigurable OFDM receiver incorporating the modulation scheme recognition feature.

The Modulation Scheme Classification component may be implemented as a normal Radio Component which attempts to estimate the modulation scheme used for each of the parallel input signal sequences but does not alter the original information. Therefore, the incoming parallel signal sequences ‘see’ a transparent signal-processing stage but the identity of the modulation schemes associated with each signal sequence is passed to the Modulation Scheme Notification stage, implemented as a Control Logic function. The Symbol De-Mapping stage (demodulator) is dependent on information from the Modulation Scheme Classification stage. This Modulation Scheme Notification stage, accessed using the Control Logic interface, passes the modulation scheme identity information to the Symbol De-Mapping stage. Using this information, the demodulator may then apply the correct symbol de-mapping protocol to each of the parallel sequences of data symbols.

The underlying structure of the reconfigurable OFDM transceiver incorporating the modulation scheme classification and Control Logic functions is different than the original transceiver stages shown in Figure 10.1. In order to accommodate the different modulation schemes that may be used for each of the parallel binary input sequences, the Serial to Parallel (S/P) stage is now placed before the Symbol Mapping stage. Regarding the receiver section of the radio, Parallel to Serial (P/S) conversion now occurs after the symbols are de-mapped to binary value sequences. This radio re-structuring is not a complicated task using the IRIS system as the initial radio configuration is defined simply, using the IRIS XML configuration description file. Dynamic re-structuring at run-time can be achieved using the IRIS Control Logic functions and interface.

10.6 Evaluation

This section presents results from a MATLAB simulation of the modulation scheme classification technique considering a communication channel affected by AWGN and random phase noise. The intercepted signal cases considered are BPSK and QPSK waveforms using one sample per symbol. Evaluation of modulation scheme classifiers may be in terms of BER, CNR or SNR. The results presented in this section focus on the calculated 8th order statistical moments versus SNR for each of the modulation scheme cases, in addition to the SNR versus BER for both signal cases. The statistical moments are calculated about the point $0.5 + 0j$ on the constellation diagram.

The graph of the output of the proposed modulation scheme classification technique versus the SNR is shown in Figure 10.4. The number of transmitted symbols is 1000 and the symbols are equi-probable. It is assumed that the receiver has achieved perfect symbol and carrier frequency synchronisation. The simulated transmitted signal is corrupted by AWGN and random phase noise, the SNR is varied from 40 dB to 0 dB and no channel equalisation processes are applied to the intercepted signals. The modulation scheme classification technique clearly distinguishes between the BPSK and QPSK cases for high SNR cases. If $M_{thresh} = 4500$, then the classifier will correctly discriminate between the two modulation schemes until the SNR is reduced to approximately 15 dB. As the SNR degrades, the classification metric reduces as deviation of signal points about the expected constellation states increases due to the increase in noise.

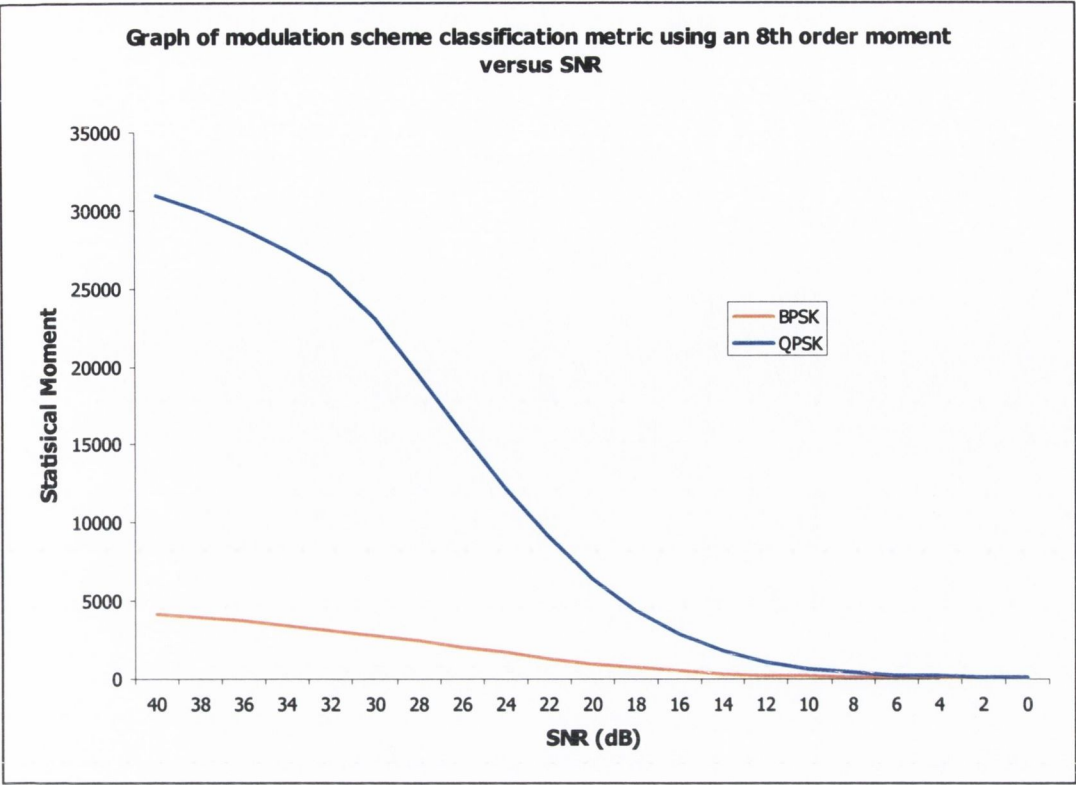


Figure 10.4: Graph of the RFCD modulation classification metric versus SNR for BPSK and QPSK cases.

Figure 10.5 is a graph of the SNR versus the BER calculated during the classification simulation described previously for the BPSK and QPSK cases. It is assumed that perfect symbol and carrier frequency synchronisation has been achieved. The transmitted waveforms are affected by AWGN and random phase noise. The graph indicates that the BER for both sets of signals remains at zero until the SNR is reduced to approximately 10 dB.

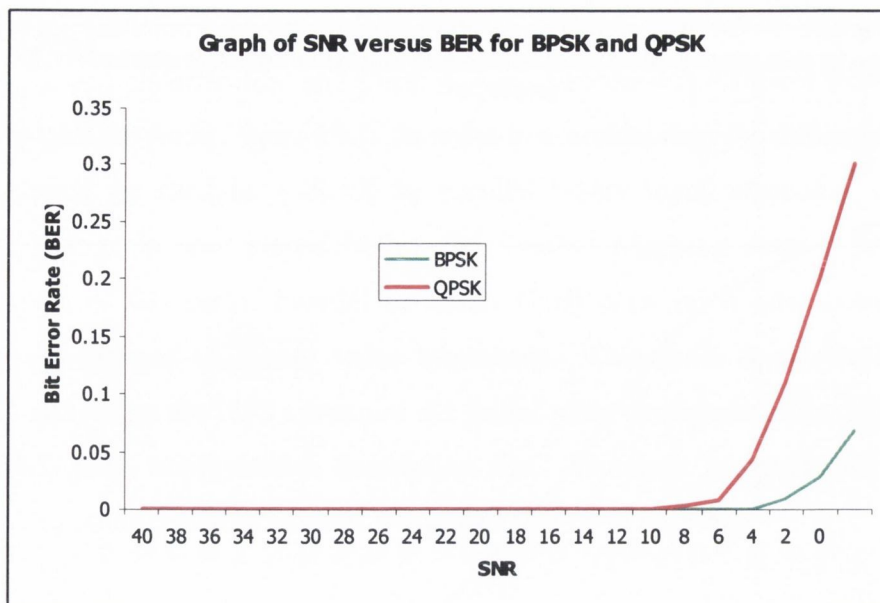


Figure 10.5: Graph of the BER versus SNR for the BPSK and QPSK classification test.

The performance of this modulation scheme classifier in a multi-path wireless channel environment is of greater interest. A reconfigurable radio may be faced with received signals that have been affected by frequency-selective fading, Doppler and multi-path Rayleigh fading effects. In order to examine the effects of this wireless environment, a MATLAB simulation is used to test this scenario.

The received signal, which may be modulated using BPSK or QPSK, comprises one direct path and three multi-path Rayleigh-fading components. The sample set of signals is 100 BPSK/QPSK equi-probable symbols and each symbol is represented as one signal sample. All of the signal components are affected by Doppler shift with a maximum frequency of 5 Hz to simulate a fast moving mobile reconfigurable OFDM receiver. The dominant direct path component has a channel gain of 3 dB and is affected by Rician fading. The first multi-path signal component, delayed by 0.2 ms is affected by Rayleigh fading and has a channel gain of -3 dB. The second multi-path Rayleigh fading signal component is delayed by 0.4 ms and has a channel gain of -12 dB. The third signal component that arrived at the receiver 0.7 ms after the direct path signal, via a propagation path affected by Rayleigh fading, has a channel gain of -30 dB. It is assumed that there is no carrier frequency or symbol timing misalignment at the receiver. This channel model is shown in Figure 9.7.

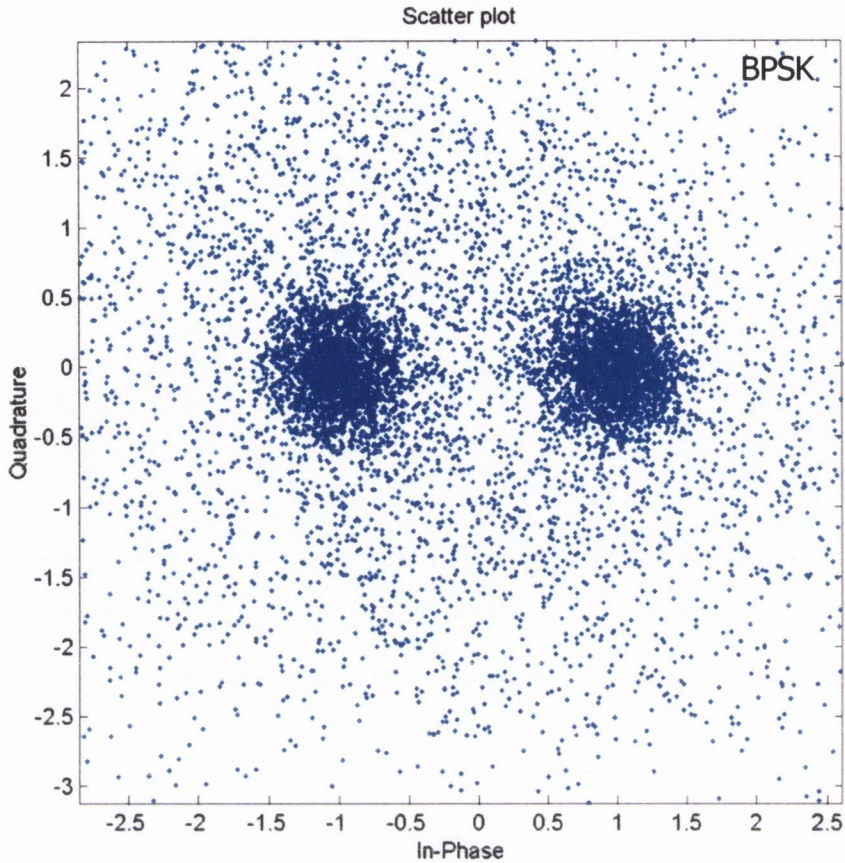


Figure 10.6: Scatterplot of a BPSK signal affected by Rician and multi-path Rayleigh fading.

Figure 10.6 is a scatterplot of a sample of the intercepted BPSK signal without receiver equalisation. This shows that the fading and multi-path signal components significantly degrade the originally transmitted signal. The measured BER for this example was 0.13, which is a significant loss. The modulation scheme classification technique reported an 8th order statistical moment of approximately 53.

Figure 10.7 is a scatterplot of a sample of the intercepted QPSK signal without receiver equalisation. This shows that the fading and multi-path signal components also significantly degrade the originally transmitted signal. The measured BER for this example was 0.35, which is an exceedingly high error rate. The modulation scheme classification technique reported an 8th order statistical moment of approximately 99.

Despite the high BER for both of these signal cases, the output of the modulation scheme classification technique for the BPSK case is significantly less than the classification metric obtained for the QPSK case. This example shows that by setting the classification threshold value to $M_{thresh} \approx 70$, in this case, it is possible to discriminate between the two modulation schemes.

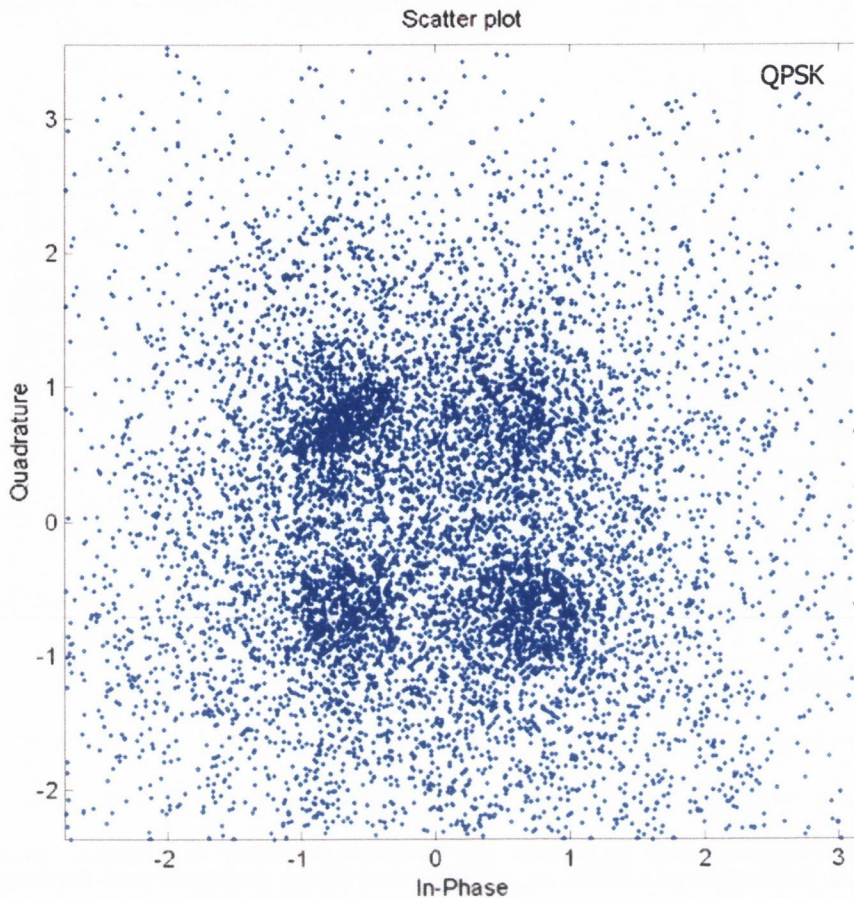


Figure 10.7: Scatterplot of a received QPSK signal affected by Rician and multi-path Rayleigh fading.

An example of how this modulation scheme classification technique may be used to create an enhanced reconfigurable OFDM transceiver is shown in Figure 10.8. The Modulation Scheme Notification stage implemented as a Control Logic function may be used to create an adaptive modulator. The estimated modulation schemes used for each sub-carrier may also be used to reconfigure the Symbol Mapping stage thus enabling the transmitter to match the modulation scheme used for incoming signal waveforms.

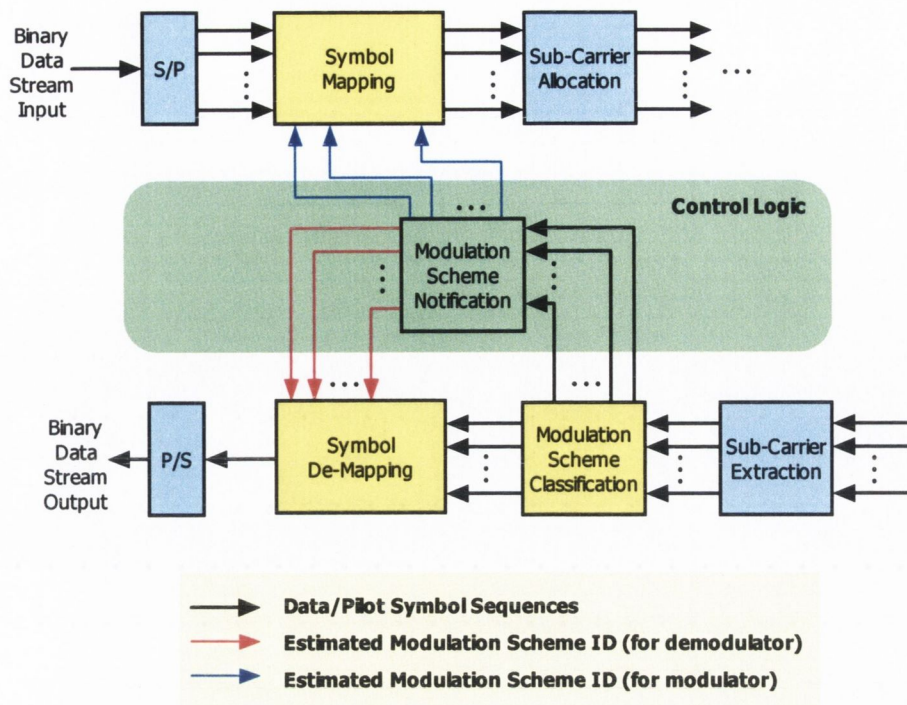


Figure 10.8: Block diagram showing how modulation scheme knowledge may be used to facilitate adaptive modulation using the IRIS framework.

10.7 Conclusion

This chapter has presented a modulation scheme classification technique for the OFDM reconfigurable radio. The proposed classification technique is a phase-based classifier that estimates the modulation scheme using the statistical moments of the received signal-points on a constellation diagram. This scheme is designed to have a low processing-power overhead by reducing the number of statistical moment calculations in half. Modulation scheme classification may increase the data-rate capabilities of OFDM by enabling each of the sub-carriers to employ different modulation schemes depending on the channel gain associated with each sub-carrier. This may be achieved by combining the modulation scheme classification technique with the sub-carrier allocation technique proposed in the previous chapter. It is also envisaged that no extra information is required to be transmitted by the source OFDM transceiver regarding the modulation scheme used for each of the sub-carriers thus the transmission overheads may also be minimised.

11

CONCLUSIONS

11.1 Introduction

This chapter presents conclusions from this thesis. Section 11.2 summarises the specific contributions of this work. Section 11.3 outlines example of how these contributions may be developed further. Section 11.4 concludes.

11.2 Summary of Contributions

The main purpose of this work has been to show that OFDM systems may be enhanced using reconfigurable radio as an enabling technology for wireless channel adaptation techniques. This thesis may be viewed as consisting of three main parts:

- A. Introduction to OFDM and related products and applications.
- B. Reconfigurable radio platform and OFDM implementation.
- C. Novel OFDM-enhancing techniques using this reconfigurable radio platform.

11.2.1 Part A

11.2.1.1 A Concise Introduction to OFDM

The first part of this thesis described the main events and key research related to OFDM technology and provides a comprehensive explanation of the processes involved in the creation of an OFDM waveform and demodulation of a received baseband OFDM signal. An overview of the main commercially-available OFDM-based products and services was also presented.

11.2.2 Part B

11.2.2.1 Reconfigurable OFDM Transceiver

The second part of this thesis presented the signal-processing platform, which forms the foundation for development and implementation of an enhanced reconfigurable OFDM system. An implementation of a reconfigurable OFDM system using a General Purpose Processor was also presented in order to create a strong foundation for the rest of the work presented in this thesis. The reconfigurable radio was created using the IRIS reconfigurable radio framework, which was designed by Mackenzie [Mackenzie2004]. The signal-chain of a radio is broken down into groups of one or more signal-processing stages. Each grouping is implemented in software to form an IRIS Radio Component. The IRIS platform enables these Radio Components to be arranged to form a radio. These Radio Components can be dynamically re-arranged, replaced and modified to form a reconfigurable radio capable of continuously adapting to the wireless communications channel environment.

This thesis provides details of how the modulation and demodulation signal-processing stages of an OFDM system may be implemented as a sequence of reconfigurable Radio Component signal-processing stages. Each of these Radio Components has been created using a generic IRIS component structure. Details of how the OFDM system may be reconfigured based on information regarding the characteristics of the wireless communications channel have also been provided. This OFDM reconfigurable radio platform is the key enabling-technology allowing the realisation of the following enhanced OFDM-based communications technique proposals.

11.2.3 Part C

The third part of this thesis presented explanations and analysis of the novel techniques developed to enhance OFDM using the reconfigurable radio platform.

11.2.3.1 Frame Synchronisation Technique

Frame synchronisation is a key stage of the OFDM demodulation process. A received group of OFDM symbols contained in an OFDM frame can only be extracted and demodulated correctly if the beginning of each OFDM frame can be estimated accurately. The low-complexity frame synchronisation technique presented in this thesis was designed for the OFDM reconfigurable radio and is a correlation-based technique. This can also be used to correct any carrier frequency offsets that may exist between the transmitter and

receiver. The results presented in this thesis showed that the start of an OFDM frame can be estimated even when the SNR is 0 dB using a single frame synchronisation training symbol.

11.2.3.2 Dynamic OFDM technique

A novel frequency-hopping OFDM technique called Dynamic OFDM (DOFDM) has also been presented. This technique enables the OFDM sub-carriers to be randomly distributed over a wider frequency range. The main features of this scheme are:

- The peak power of the OFDM transmission signal is reduced by over 10 dB thus reducing the possibility of non-linear PA operation, signal distortion and spurious emissions.
- Security of transmitted information against possible eavesdroppers is increased as the transmitted information is scrambled.
- The robustness of OFDM against multi-path fading, interference and noise is increased further as sub-carriers are used for only a very short length of time and the frequency-diversity properties of OFDM are enhanced further.
- The frequency-hopping sequences do not need to be conveyed to the remote OFDM receivers thus the transmission-overhead of this system is minimised.
- The proposed DOFDM technique uses the proposed frame synchronisation technique as a means of synchronising the OFDM receiver with the frequency-hopping cycle of operation thus increasing the efficiency and versatility of both the DOFDM and frame synchronisation schemes.

11.2.3.3 Sub-carrier Allocation Technique

A traditional fixed-architecture OFDM transceiver uses pre-determined sub-carrier frequencies regardless of the interference and noise that may be present on one or more of these frequencies. A novel sub-carrier allocation technique has been presented that enables sub-carriers with excessive noise and/or interference to be avoided by the OFDM transmitter using a periodogram threshold-based algorithm. This technique increases the

robustness of OFDM by reducing the number of bit corrections required in the received data stream or just simply blocked due to the noise/interference present on the carrier frequency.

The second main contribution of this sub-carrier allocation technique is the method used to inform the receiver which sub-carriers are being used to transmit information from the source during each OFDM frame. This is achieved using the frame synchronisation training symbol. Each of the possible sub-carriers is represented by one binary value; a binary one indicates that that particular sub-carrier is being used to transmit information and a binary zero is used to indicate that a sub-carrier will not be used. All of the binary values representing the total set of sub-carriers encapsulated in an array to form a channel mask. This channel mask is then modulated to form a sequence of complex-valued pilot symbols. These pilot symbols are then used to form an enhanced frame synchronisation and carrier-frequency offset estimation technique. Upon estimating where the start of an OFDM occurs in an intercepted OFDM signal, the frame synchronisation training symbol is demodulated. This yields the binary values designating the sub-carriers being used to carrying information in that current OFDM frame. Results for a multi-path Rayleigh fading, direct path Rician fading, Doppler fading and AWGN combined channel model indicated that a BER of 0.2 for a series of received channel masks was achievable for a channel SNR of 20 dB.

The final results set for the sub-carrier allocation technique involved comparing the BER and data-throughput of normal fixed sub-carrier OFDM and the dynamically-allocated enhanced OFDM technique over a simulated wireless channel model affected by strong multi-frequency interference. It was shown that the dynamically-allocated sub-carrier OFDM technique outperforms the normal fixed sub-carrier OFDM. Compared to a 48 sub-carrier fixed allocation OFDM signal, which had a BER of approximately 0.015 where $E_b/N_0 \geq 8$, and a total uncorrected data throughput of 96000 bits per OFDM frame, the dynamically-allocated interference-avoidance OFDM technique achieved a BER of 5.5×10^{-6} for $E_b/N_0 \approx 20$. The total number of uncorrected received data per frame using this technique was significantly increased to 180000 bits.

11.2.3.4 Modulation Scheme Recognition Technique

A novel low-complexity modulation scheme recognition technique designed for reconfigurable radio implementation has also been presented. This is a phase and

statistical moment based classifier with an objective of presenting a low processing-time overhead to the reconfigurable radio receiver process. Results for a multi-path Rayleigh fading, direct path Rician fading, Doppler fading and AWGN combined channel model indicated that this proposed classification technique can distinguish between non coherent BPSK and QPSK even when the degradation in channel quality results in BER values of 0.13 for the BPSK and 0.35 for the QPSK case.

11.3 Future Work

This thesis has identified a selection of the many areas in which OFDM-based communications may be enhanced. These techniques may be improved further by testing 'live' implementations of the reconfigurable radio in a real multi-path fading and noisy channel environment. A suggested band of interest is the 2.4 GHz ISM frequency band.

11.3.1 Multi-user OFDM Wireless Bridge

This thesis examined some of the many areas in which the OFDM communications process can be improved by taking a similar reconfiguration approach. Many other aspects OFDM-related communications systems may also be improved through the use of reconfigurable radio. One future reconfigurable OFDM radio application is to create a multi-user OFDM radio server that acts as a wireless bridge between two fixed Ethernet networks. This is essentially an 'end to end IP' application using reconfigurable OFDM as the wireless bridging technology. Several different information sources may then be multiplexed as a sequence of OFDM symbols and transmitted to the destination transceiver. At the stage, the received multiplexes of independent information streams are de-multiplexed to re-join the fixed Ethernet network once again. Reconfigurable radio technology enables the novel sub-carrier allocation technique presented in this thesis to be enhanced further to enable the available sub-carriers used for transmission to be allocated to users based on the priority of the service, number of users already using the spectrum and the wireless channel characteristics associated with each of the sub-carriers.

Details of one scenario using this proposed scheme is called 'Multiple User Data Enhanced Radio Server (MUDERS)' and is published in proceedings of the 4th Software Defined Radio (SDR) Forum Technical Conference 2004 [Nolan2004].

11.3.2 Cognitive Radio

It was argued in this thesis that in order to create a radio capable of adapting to the user's requirements, one of the initial main challenges is creating a radio that can adapt to the wireless channel. The techniques involved in the wireless adaptation process form the basis of higher-level user adaptation features. Therefore the work presented in this thesis may be developed further to implement a version of a cognitive radio.

A cognitive radio [Mitola1999] as defined in Section 4.3.5 may be created by increasing the freedom of reconfigurability offered by the reconfigurable radio implementation presented in this thesis to aid the creation of an autonomous wireless device. The enhanced reconfigurable OFDM system presented in this thesis can be developed as a basis for cognitive radio as it provides the necessary separation between the air-interface, wireless service and user's preferences required for a cognitive radio application. The capabilities of the reconfigurable radio can be extended to allow extra information relating to the current location of the device, external temperature, available power, usage-history patterns and user-preferences to influence the way the radio accesses the wireless medium.

11.3.3 Spectrum Management and Spectrum Sharing

Spectrum-sharing [Oner2003] [Weiss2004][Leaves2004] is an rapidly developing technology where frequency-spectrum is allocated to users on a temporary or time-sharing basis in order to increase spectrum-usage efficiency and promote fair co-existence between the numerous wireless services accessing a spectrum segment. Reconfigurable radio and enhanced OFDM can be viewed as key facilitating technologies for this system.

Current RF spectrum regulators including the Commission for Communications Regulation (COMREG) [COMREG] in Ireland and the Office of Communications (OFCOM) [OFCOM] in the United Kingdom allocate spectrum segments to users on a mainly exclusive-use basis. These spectrum allocations may remain unused at certain times of the day by the authorised users yet are still off-limits to other potential users. Current trends in spectrum allocation policies are gearing towards opening up spectrum segments for use by other users on a shared/co-operative basis in order to exploit the potential spare spectral bandwidth.

Spectrum-usage can be viewed in three main planes, temporal, frequency and geographical location. Regarding the temporal plane, a certain band segment may experience little or no activity during the hours of darkness for example. This means that other services can potentially access this spectrum for their own requirements during this time. Spectrum-usage can also be frequency-dependent as authorised services may only use a proportion of their allocated band segment. It is possible that other services could access the remaining bandwidth without interfering with the main authorised service. Geographical location also plays a part in measuring spectrum-usage as two or more accessing the same spectrum segment but unable to detect each other can operate without interference [Mangold2004].

Neglecting the spectrum regulation issues, in order to implement spectrum-sharing and spectrum-management schemes to increase the efficiency of current spectrum-usage activity, radio resource management is the key challenge. Reconfigurable radio is an attractive facilitator for a frequency-agile radio service that can take advantage of unused spectrum segments. The ability of an autonomous reconfigurable radio to control the methods used to access the wireless medium using information from an external source means that a reconfigurable radio may be controlled by an external control entity. Reconfigurable radio can also be controlled remotely using fixed or wireless link, or even allowed autonomous control.

The ability to control and influence the operation of a remotely-located reconfigurable radio using only a wireless connection means that the power and frequencies used by the radio may be controlled. Dynamic spectrum-management is an application that could feasibly control the power output and frequency range of a reconfigurable device within range of a spectrum-management control transmitter. By managing the frequency spectrum in this manner, interference to other legitimate users in the local area could be reduced by reducing the power of the offending reconfigurable radios.

Robert and Reed [Robert2001] examined the use of software radio as a means of enabling several different radio services to co-exist on unlicensed band segments. They highlight the need for a service broker is to manage the negotiation between facilitating user requirements and wireless device configuration. Regarding this topic of End to End Reconfigurability (E2R) [E2R] [Moessner2004], one of the research areas being pursued is a suitable radio architecture required to facilitate distributed radio reconfiguration and control mechanisms. Gultchev, Moessner and Tafazolli [Gultchev2004] define a Radio

Management Architecture (RMA) that enables negotiations between the wireless terminal and authorized network control entities to deliver a radio node that is capable of communicating with the prevalent wireless system in a heterogeneous network scenario. One of the important issues addressed by this RMA is the ability to record the current state of the reconfigurable radio for rapid service restoration in the case of a malfunction, power supply failure or other eventualities. The area of vehicular communications is another area of research that is adopting reconfigurable radio principles to establish and maintain wireless communications based on dynamic spectrum allocation and management. The Dynamic Radio for IP-Service in Vehicular Environments (DRIVE) [DRIVE] and OverDRIVE [OverDRIVE] consortium-led project is directed towards establish a more flexible mechanism for fair spectrum usage among co-existing radio services. The dynamic spectrum allocation procedure in this case is based on periodic wireless service load measurements that can then be used to update the current and anticipated spectrum usage provisioning. A spectrum-broker is used to manage and allocate the frequency segments to the Radio Access Networks. Simulations showed that the spectrum-usage efficiency can be increased by approximately 30% [DRIVE10].

11.4 Conclusions

This thesis has proven that OFDM may be improved by progressing from traditional fixed-architecture radio designs to more ‘organic’ flexible-architecture radio using a reconfigurable radio platform. In addition to this shift in radio design approach, the work in this thesis has shown that a radio is no longer a static device but has the potential to evolve, adapt and react to changes in its electromagnetic view of the world. This has been made possible by developing new ways of improving OFDM using a reconfigurable radio platform. This thesis has presented an enhanced OFDM reconfigurable radio that is capable of reacting to, and adapting to changing wireless channel conditions and user requirements.

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