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Development of a Quality of Service Framework for Multimedia Streaming Applications

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Development of a Quality of Service Framework for Multimedia Streaming Applications

by

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Abstract

By the year 2012, it is expected that the majority of all Internet traffic will be video content. Coupled with this is the increasing availability of Wireless Local Area Networks (WLANs) due to their ease of deployment, flexibility and reducing roll out costs. Unfortunately the contention based access mechanism utilised by IEEE 802.11 WLANs does not suit the non-uniform or bursty bandwidth profile of a video stream which can lead to a reduced quality of service (QoS) being experienced by the end-user.

In 2005, the IEEE 802.11e protocol was ratified in an attempt to solve this emerging problem. It provides for an access prioritization mechanism based upon four separate traffic classes or access categories (ACs). Each AC is characterised by a set of access parameters that determine its level of access priority which is turn determines the amount of bandwidth available to it.

Computer simulation studies have shown that AC prioritisation can yield significant improvements in the QoS delivered over a WLAN. However, these studies have been based upon the use of static access parameters for the ACs. In practice, this is not a viable solution owing to the dynamic and unpredictable nature of the operating conditions on WLANs.

In this thesis, an experimental study of AC prioritisation based upon adaptive tuning of the access parameters is presented. This new approach to bandwidth provisioning for video streaming is shown to yield significant improvements in the QoS under a wide range of different operating conditions. For example, it is shown that by adaptively tuning the access control parameters in response to the network conditions, the number of video frames delivered that satisfy QoS requirements is more than doubled.

Declaration

I certify that this thesis which I now submit for examination for the award of Master of Philosophy, is entirely my own work and has not been taken from the work of others save and to the extent that such work has been cited and acknowledged within the text of my own work.

This thesis was prepared according to the regulations for postgraduate study by research of the Dublin Institute of Technology and has not been submitted in whole or in part for an award in another Institute or University.

The work reported on in this thesis conforms to the principles and requirements of the Institute's guidelines for ethics in research.

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Signature____ Date________________

Candidate

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Table of Contents

Table of Contents 1

List of Figures 4

List of Tables 6

Abbreviations & Acronyms 7

CHAPTER 1 INTRODUCTION 12

1.1 Problem Statement 15

1.2 Overview of Solution Approach 15

1.3 Summary of Results and Contributions 16

1.4 Thesis Outline 17

1.5 Publication Arising From This Work 18

CHAPTER 2 TECHNICAL BACKGROUND 19

2.1 Video 19

2.2 The Digital Sensor 22

2.3 Codecs 23

2.4 Introduction to Wireless Networking 26

2.5 Wireless Standards 27

2.6 Medium Access Mechanisms 29

2.6 Video Streaming 33

2.8 Video Quality of Service 35

2.8.1 Quality Metrics 37

2.9 Challenges Associated with Video Streaming over Wireless Networks 38

2.10 Chapter Summary 40

CHAPTER 3 LITERATURE REVIEW 42

3.1 Video Transmission 42

3.2 Quality Evaluation 47

3.3 State of the Art 51

3.4 Chapter Summary...52

CHAPTER 4 EXPERIMENTAL PROCEDURE 54

4.1 Scope of This Thesis 54

4.2 Experimental Testbed and Tools...55

4.2.0 Cisco QAP Interface 59

4.2.1 CNRI WRRC Application 61

4.2.2 Video File Preparation 69

4.3 Objective Video Metrics 70

4.3.1 PSNR Calculation 70

4.3.2 Video Quality Metric (VQM) 70

4.4 Scenario 1 - Analysis of Video Streamed Over Wired Connection...............74

4.5 Test Scenario 2 - Experimental Setup..75

4.6 Test Scenario 3 - Experimental Setup..78

4.7 Test Scenario 4 – Experimental Setup 80

CHAPTER 5 RESULTS & ANALYSIS 81

5.1 Experiment 1 – Video Streaming Over a Wired Network 81

5.2 Experiment 2 - Comparison of Static IEEE 802.11e Versus Dynamic IEEE 802.11e Networks for Video Streaming With A Single Client.................................85

5.3 Experiment 3 - Comparison of Static IEEE 802.11e and Dynamic IEEE 802.11e Networks for Video Streaming With Multiple Clients 89

5.4 Experiment 4 - Effect of Capacity Allocation On Video Stream Quality Over Dynamically Controlled IEEE 802.11e Networks 92

5.5 Summary 98

CHAPTER 6 SUMMARY and CONCLUSIONS 100

6.1 Summary 100

6.2 Further Work 103

REFERENCES 105

APPENDIX 112

List of Figures

Figure 1: An RGB Image split into its component parts....21

Figure 2: Pattern used for Colour Filter Array....23

Figure 3: I, P, B frame display sequence. 26

Figure 4: Backoff Procedure....30

Figure 5: 802.11e Access Categories [9a] 31

Figure 6: EDCF Access Mechanism [9] 33

Figure 7: Examples of corrupt video frames yielding the same PSNR....36

Figure 8: Video Flow Process....58

Figure 9: Cisco QAP policy map. 59

Figure 10: Cisco QAP EDCA settings....60

Figure 11: WRRC Operation 62

Figure 12: WRRC Configuration Settings Window 63

Figure 13: WRRC Minimum Capacities Configuration Window 64

Figure 14: WRRC Main Window....65

Figure 15: WRRC main window showing Video panel 67

Figure 16: WRRC main window showing EDCA panel 68

Figure 17: Modified VQM Weighting Curve 72

Figure 18: Network topology used for experimental scenario 2....76

Figure 19: Example of results generated showing PSNR PDF, PSNR CCDF, VQM PDF and VQM CCDF....78

Figure 20: Network topology used for experimental scenario 3....79

Figure 21: Expected PDF of PSNR values 81

Figure 22: PDF of PSNR values for video streamed on a wired connection....82

Figure 23: PSNR values Little Miss Sunshine varying resolution and bitrate....84

Figure 24: CDF of VQM Miss Sunshine Topology 1 Dynamic EDCA vs. Static EDCA 86
Figure 25: CDF of VQM Scanner Darkly Topology 1 Dynamic EDCA vs. Static EDCA 88
Figure 26: CDF of VQM Miss Sunshine Topology 2 Dynamic EDCA vs. Static EDCA 90
Figure 27: CDF of VQM Scanner Darkly topology 2 dynamic EDCA vs. static EDCA 91
Figure 28: Effect of capacity allocation: Scanner Darkly 2500kbps Background 750pps 768B 92
Figure 29: Effect of Capacity allocation: Scanner Darkly 2500kbps Background 750pps 256B 93
Figure 30:CDF of VQM Miss Sunshine 2000kbps Background 750pps 1024B Topology 1..94
Figure 31: CDF of VQM Miss Sunshine 2000kbps Background 750pps 1024B Topology 2 96
Figure 32: Video Quality vs. Capacity Miss Sunshine 2000kbps Background 70pps 1024B 97

List of Tables

Table 1: RGB Pixel values and corresponding colours 20

Table 2: QoS values used by Ksentini, Gueroui and Naimi 42

Table 3: EDCA values used. (Ksentini et al.) 43

Table 4: DCF parameters. (Ksentini et al.) 43

Table 5: Parameters of the 802.11e ACs (Shin and Chung) 44

Table 6: Mapping of video layers to ACs for four different schemes 45

Table 7: EDCA values for each AC (MacKensie et al.) 46

Table 8: EDCA settings used for varying *CWmin* experiment 47

Table 9: EDCA settings used for varying *AIFSN* experiment 47

Table 10: Traffic types and characteristics used (Gao et al.) 48

Table 8: Static 802.11e and Default 802.11e settings 54

Table 9: Video Characteristics for wired connection 74

Table 10: Best Effort Traffic Levels 76

Table 11: Video files, Background Loads and Topologies used 80

Table 12: Improvement in P[VQM < 0.5] 98

Table 13: Variation in P[VQM < 0.5] with packet size. 99

Table 14: Variation in P[VQM < 0.5] with topology. 99

Abbreviations & Acronyms

3G	Third Generation
AC	Access Category
AEF	Access Efficiency Factor
AIFS	Arbitration Interframe Space
AP	Access Point
ATSC	Advanced Television Systems Committee
AP	Access Point
BBC	British Broadcasting Corporation
bps	bits per second
BW	Bandwidth
CBR	Constant Bit Rate
CCDF	Complimentary Cumulative Distribution Function
CDF	Cumulative Distribution Function
CDMA	Code Division Multiple Access
CMYK	Cyan Magenta Yellow Black
CNRI	Communications Network Research Institute
CIF	Common Intermediate Format
CoS	Class of Service
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CW	Contention Window

dB	Decibel
DCF	Distributed Coordination Function
DIFS	DCF Interframe Space
DVB	Digital Video Broadcasting
DVD	Digital Versatile Disc / Digital Video Disc
EDCA	Enhanced Distributed Channel Access
EDCF	Enhanced Distributed Coordination Function
ES	Elementary Stream
EU	European Union
fps	Frames Per Second
GHz	Gigahertz
GOP	Group of Pictures
HCCA	HCF Controlled Channel Access
HCF	Hybrid Coordination Function
HD	High Definition
HVS	Human Visual System
Hz	Hertz
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Propocol

ISDB	Integrated Services Digital Broadcasting
ISM	Industrial, Scientific, Medical
ITS	Institute for Telecommunication Sciences
JPEG	Joint Photographic Experts Group
kbps	Kilobits per second
LAN	Local Area Network
LCD	Liquid Crystal Display
MAC	Media Access Control
Mbps	Magabits per second
MIMO	Multiple Input Multiple Output
MOS	Mean Opinion Score
MPEG	Moving Picture Experts Group
MSDU	MAC Service Data Unit
MSE	Mean Square Error
MTU	Maximum Transmission Unit
NTSC	National Television System Committee
PAL	Phase Altering Line
PCF	Point Coordination Function
PDF	Probability Density Function

PES	Packetized Elementary Stream
PHY	Physical Layer
PPS	Packets Per Second
PSNR	Peak Signal to Noise Ratio
QAP	QoS enabled Access Point
QCIF	Quarter Common Intermediate Format
QoD	Quality of Delivery
QoE	Quality of Experience
QoS	Quality of Service
RGB	Red Green Blue
RTE	Radio Telefis Eireann
SD	Standard Definition
SECAM	Sequential Colour with Memory
SIFS	Short Interframe Space
SSID	Service Set Identifier
STA	Wireless Station
TV	Television
TXOP	Transmission Opportunity
USB	Universal Serial Bus

VBR	Variable Bit Rate
VoD	Video on Demand
VQM	Video Quality Metric
WLAN	Wireless Local Area Network
WMM	Wi-Fi Multimedia
WMN	Wireless Mesh Network
WRRC	Wireless Radio Resource Controller
YUV	Luma Component (Y) Chrominance Components (UV)

CHAPTER 1 INTRODUCTION

Video traffic on global Internet Protocol (IP) networks is estimated to account for 40% of all traffic by the end of 2010 according to a recent Cisco market report [1]. Video traffic places large demands on a network in terms of packet loss, delay, jitter and bandwidth that quickly exposes any weaknesses in the network. Furthermore, if a wireless network is employed, video streaming becomes an even more challenging task.

Traditional IEEE 802.3 or Ethernet wired Local Area Networks (LANs) can reliably offer data rates of up to 1 Gbps. Despite this, Wireless Local Areas Network (WLAN) roll out continues to grow since its inception due to a number of factors, namely:

Ease of installation – WLANs don’t require large amounts of cable to be run between stations.

Mobility – Users are not restricted to where they can access the network, provided they are within the transmission range, leading to increased collaboration within workforces.

Flexibility – Upgrading and extending wireless networks is a less laborious task compared to wired networks. Typical transmission range is approximately 100m which is adequate for residential applications and small to medium sized businesses.

Cost – Like most technology, as wireless networking matures, the costs involved for infrastructure are decreasing.

Unfortunately the wireless network does not provide the perfect alternative to traditional wired networks. WLANs offer a considerably reduced bandwidth and can therefore transmit less data per second. Interference from external sources and channel fading can corrupt transmissions on a wireless network and lead to unacceptable packet loss levels. Wireless networks also present new security challenges for administrators not usually associated with wired networks.

The popularity of video streaming applications has led to an increase in the volume of video traffic being transmitted over the 3G CDMA wireless network. In areas where traditional broadband services are difficult to install or where mobile Internet access is required, 3G wireless USB adapters provide an alternative means of Internet access. Many 3G mobile and smart phones now have the ability to stream video content from popular video hosting websites and also the ability to perform video calling.

Video content places a large bandwidth demand on the 3G CDMA network and in an effort to reserve the bandwidth required for voice call content and other data applications mobile operators are investigating the use of *data offload* [2].

Data offload aims to use a multi-protocol approach to deliver content using both the 3G CDMA network and traditional IEEE 802.11 WLANs. The goal is to utilise IEEE 802.11 networks for high bandwidth demand applications and the 3G CDMA network for other less demanding applications.

Data offload places further video traffic on the IP network infrastructure and specifically on the wireless IEEE 802.11 networks.

With the exception of the IEEE 802.11e protocol, WLANs only offer a best effort system that does not distinguish between data and real time traffic such as voice and video. The real time nature of these applications places strict delay requirements on the network as data arriving after its play-out time is usually regarded as lost. In comparison, other forms of data traffic can suffer longer delays without being apparent to an end user. Video traffic requires large volumes of data to be transmitted per second and therefore has a relatively high bandwidth demand compared to other applications.

The real time, high bandwidth characteristic of video traffic does not suit the traditional best effort, limited bandwidth and high loss rate characteristic of traditional WLANs.
In an effort to give more priority to real time data on a WLAN, the IEEE developed the IEEE 802.11e protocol. This protocol provides for four separate traffic queues, called Access Categories (ACs), on the network. Each AC can be given different access opportunities to the wireless network. The ACs are labelled *Voice*, *Video*, *Background* and *Best Effort*. By modifying a set of parameters, known as EDCA parameters, relating to a transmission queue it can be given a higher probability of winning access opportunities leading to more bandwidth being available to transmit its load. A problem with the IEEE 802.11e protocol is that there are many parameters (i.e. four EDCA parameters for each AC) that need to be set by the network administrator. In order to take full advantage of the protocol not only do these parameters need to be set appropriately but they also need to be set dynamically according to the traffic load conditions present on the network. To accomplish this task a Radio Resource Management (RRM) tool needs to be employed.

Determining the quality of service (QoS) of a video stream that has been transmitted across a network is not a trivial task. Details of the transmission itself, for example: delay, jitter, loss rate and bandwidth, can be obtained using conventional networking tools and are labelled Quality of Delivery (QoD) metrics [3]. The quality of the video as perceived by the end user, known as the Quality of Experience (QoE), does not directly relate to the QoD observed for a stream. In order to accurately determine QoE, time consuming and costly subjective live human trials can be used. Due to the limitations of live trials, objective quality metrics were developed to estimate the quality that would be experienced by an end user. Of these metrics the Peak Signal to Noise Ratio (PSNR) has been widely employed but it does have some disadvantages as it does not weight corruptions according to their visibility and therefore has a poor correlation to the Human Visual System (HVS). The Video Quality Metric (VQM) expands upon the PSNR metric in order to weight the degradations to reflect the end users experience more accurately.

1.1 Problem Statement

Video streaming over conventional IEEE 802.11 wireless LANs is a challenging task. The video stream itself places large bandwidth and low packet loss requirements on the limited bandwidth hostile wireless network. It is also a bursty application that does not suit the contention mechanism employed by IEEE 802.11 WLANs. These characteristics can lead to poor video quality being experienced by the end user. This thesis presents an experimental investigation carried out to understand the relationship between the proportion of video frames delivered that meet user requirements and the minimum capacity allocated to the video AC. This experimental approach involves streaming real video over an IEEE 802.11e network where the CNRI WRRC [4] provides an RRM system that dynamically tunes the EDCA settings of the IEEE 802.11e protocol based on the current network load conditions and a set of user specified Minimum Capacity (*Cmin*) values.

1.2 Overview of Solution Approach

Throughout the work documented in this thesis, video traffic was transmitted between a host and client side machine for a range of topologies and background traffic levels. The transmitted video files are also varied in terms of bit rate, resolution and visual content.

Using the PSNR and a modified version of the VQM, the proportion of video frames that met end user quality requirements at the client side was determined.

By employing dynamic control of the EDCA parameters the proportion of frames that met end user quality requirements has been more than doubled when compared to the use of static EDCA parameters.

1.3 Summary of Results and Contributions

Previous work in the field has been based on computer simulation where the NS-2 simulation package has been used. This thesis is based upon an experimental study where real time RRM has been implemented. As the level of video traffic continues to increase there is an urgent need for RRM in order to provide the most effective use of the limited availability of bandwidth on WLANs to deliver video services [23].

The main findings of this work are that static EDCA settings do not work well in practice due to the dynamic characteristics of the wireless medium. Other than trial and error, it is not known how to set these values to deliver video content that satisfies end user quality requirements. To take full advantage of the IEEE 802.11e protocol the EDCA settings need to be adaptively tuned in response to background traffic loads. It has been shown experimentally that the bandwidth required by a video clip is not solely dependent on the encoding configuration used but also strongly depends on the visual complexity of the content [44].

This thesis contributes to the body of work in the field by providing results based upon an experimental study using video clips, rather than computer simulation using video trace files. In particular, the issue of determining how much bandwidth should be allocated to a video stream to preserve video quality is investigated. In addition, this study has generated a set of PDF and CDF distributions for the video quality which allows for statistical or “soft” QoS provisioning to be implemented. These distributions can be used by network operators to trade-off bandwidth for QoS in order to provide the minimum acceptable video quality to the largest volume of customers, therefore maximising the use of their network resources.

Unlike other approaches that involve partitioning a video into a number of ACs, this approach preserves the MPEG-4 frame structure and transmits a video stream through a single AC. This avoids the complexity required to manage the transmission of a video through multiple ACs. The use of the CNRI WRRC also reduces the number of variables that need to be set for each AC from four to one, namely the *Cmin* setting.

1.4 Thesis Outline

Chapter 2 provides background information on digital video formats and standards and provides data relating to the various wireless networking standards.

Chapter 3 provides a review of the relevant published work in the area of video streaming, QoS provisioning and QoE evaluation.

Chapter 4 describes the four experimental scenarios examined and details the tools and testbeds required to generate results.

Chapter 5 provides the results obtained in the various experimental scenarios.

Chapter 6 details a final summary of the work undertaken. Conclusions and suggested areas of possible future work are also offered.

1.5 Publication Arising From This Work

Stuart Wallace and Mark Davis 2008 *Effects of Line Rate on Video QoS over Wireless Networks – an Experimental Approach*, Information Technology and Telecommunications Conference (ITT 2008), Galway Mayo Institute of Technology, Ireland.

CHAPTER 2 TECHNICAL BACKGROUND

2.1 Video

In its simplest form a video is a collection of sequential still images displayed one after another with a set frequency to give the appearance of motion. Primarily video display standards were developed for analog television services. Many are still used today although the EU has mandated that all EU countries have switched to digital services by 2012 [5].

The dominant analog television display standards are PAL, NTSC and SECAM. PAL is the standard used in the majority of European countries, Australia and parts of Asia. NTSC is used in North and South America and Japan while the SECAM standard is used in Russia, France and many African countries. These standards are gradually being phased out and replaced with digital standards: DVB in Europe, ATSC in North America and ISDB in Japan.

The fundamental parameters of a digital video are frame rate, resolution, bit rate and aspect ratio.

Frame rate relates to the number of still images or *frames* to display per second (fps). PAL and SECAM systems use a frame rate of 25fps while NTSC systems display at 29.97fps. The ISDB digital standard supports 30fps while ATSC and DVB both support frame rates up to 60fps.

The resolution of a digital image represents the physical dimensions of the image area expressed in terms of pixel width and pixel height. A pixel is the smallest discrete component of a digital image and represents a single point of an image. A pixel represents the intensity of certain chosen colours for that point of an image. Different colour combinations may be used for different applications but all versions are defined as Colour Spaces. RGB colour space may be the most recognised but several other forms exist including YUV, CMYK and LAB.

In RGB colour space each pixel comprises three values in the range 0 to 255 corresponding to the intensity of **Red**, **Green** and **Blue** respectively. This combination allows for over 16 million individual colours to be rendered. Table 1 below illustrates how some colours are represented. Figure 1 displays an RGB image displayed as its individual colour components. Each channel represents the intensity of that colour where darker grey represents more intense colour.

A resolution of 640 pixels x 480 pixels represents a horizontal image size of 640 pixels and a vertical image size of 480 pixels.

Red	**Green**	**Blue**	**Resulting Colour**
0	0	0	Black
255	255	255	White
128	128	128	Medium Grey
255	0	0	Red
0	255	0	Green
0	0	255	Blue

Table 1: RGB Pixel values and corresponding colours

(a) RGB Image

(b) Red Channel

(C) Green Channel

(d) Blue Channel

Figure 1: An RGB Image split into its component parts

The bit rate of a digital video describes the volume of data consumed per unit time. It is measured in bits per second (bps), but is more commonly expressed as kbps. Bit rate can be either constant over time (CBR) or variable over time (VBR). CBR video encoding uses the same amount of data over a given time regardless of the video's visual complexity. VBR encoding adapts to the video complexity and can use less or more data as required to maintain a given compression/quality requirement. The decision to use one form over another is influenced by the application, quality constraints, network constraints and content.

The aspect ratio refers to the ratio of the width of the video frame to the height of the video frame. Standard definition video uses an aspect ratio of 4:3 while widescreen videos and High Definition TV employs a 16:9 ratio. As standard definition services are removed the 16:9 ratio is becoming increasingly common.

2.2 The Digital Sensor

Digital imaging sensors convert the analog light levels of a scene into an electrical charge by using an array of silicon devices called photodiodes. The resolution of a digital sensor is expressed as the amount of pixels it can record, usually expressed as megapixels. The charge recorded by each photodiode is then amplified and digitised into pixel data. These photodiodes cannot distinguish colour, they can only record the accumulated intensity of light. For this reason, to record data relating to colour a filter must be used in order to split normal light into red, green and blue light. Typically twice the amount of data relating to green than either red or blue is captured [6]. More green data is captured as the human eye is more sensitive to green light than red or blue light. One pixel is composed of 4 photodiodes; 2 green, 1 red and 1 blue. The colour filter array most commonly used is laid out in the pattern shown in Figure 2.2.

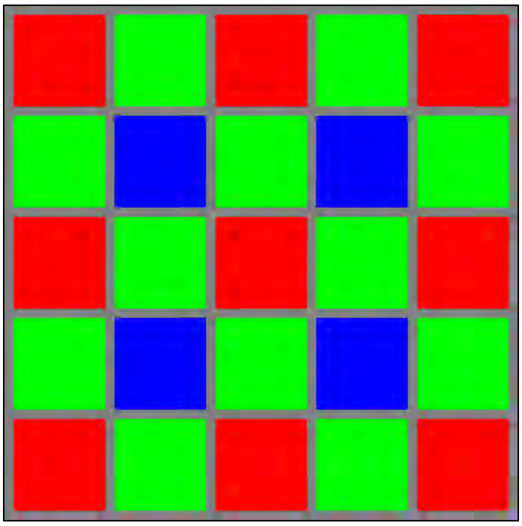

Figure 2: Pattern used for Colour Filter Array

Unprocessed digital video data streams are generally unnecessarily large for every day use and place vast demands on processors, storage devices and networks. To reduce these demands, video is generally processed using a codec (Coder Decoder). This system reduces file size by employing compression techniques to remove imperceptible details and artefacts. The International Standards organisation (ISO) is responsible for the standardisation of several codecs. The more popular codec standards are MPEG-1, MPEG-2, MPEG-4 and H.264. Other proprietary codecs have been developed by software companies. For example Microsoft developed Windows Media Video (WMV) and Apple have developed the Quicktime Movie format (MOV).

2.3 Codecs

MPEG-1

The MPEG-1 standard consists of 5 parts; Systems, Video, Audio, Compliance Testing and Software Simulation.

The systems section covers the problem of combining one or more data streams from the video and audio parts of the MPEG-1 standard with timing information to form a single stream. Part 2 specifies a coded representation that can be used for compressing video sequences to bitrates about 1.5Mbps. Part 3 specifies how audio streams should be compressed and is well known as it is employed by mp3 audio conversion methods.

MPEG-2

MPEG-2 is composed of 9 parts; the first 3 are systems, video and audio. Part 1, systems, describes how to multiplex video and audio synchronously. This is specified in two forms: Transport Stream and Program Stream.

The program stream is similar to the MPEG-1 multiplex and combines one or more Packetized Elementary Streams (PESs) which have the same time bases into a single stream. An elementary stream (ES) is the output stream from an encoder, either: video, audio or closed caption. The elementary stream is then packetized by encapsulating sequential data bytes from the elementary stream inside PES packet headers. It is designed for use in relatively error-free environments such as optical discs and is used as the compression method for standard DVD videos.

The transport stream is designed for use in error prone environments and combines one or more PESs which have independent time bases into a single stream.

Part 3 is a multi-channel, backwards compatible, extension to MPEG-1 audio.

MPEG-4

The MPEG-4 standard was ratified in early 1999. It builds on the success of three fields: Digital Television, Interactive Graphics Applications and Interactive Multimedia.

MPEG-4 enables the production of content that has greater reusability, flexibility and provides content developers with a means to protect their work. For end users it brings higher levels of interaction and the ability to bring multimedia to new networks such as mobile networks.

MPEG-4 encoding utilizes three different frame types for video composition. These are *Intra Coded* frames (I-Frames), *Predicted* frames (P-Frames) and *Bidirectional Coded* frames (B-Frames).

I-Frames are encoded as JPEG images. They contain data for each pixel of the image and are independent of past or future frames in the stream. P-Frames contain data relating to what has changed in the scene since the last reference frame, either I or P. This is calculated on a block by block basis. B-Frames contain data based upon changes that have happened since the last frames and also on the changes that will occur in the subsequent frames of the stream.

I-Frames contain the most image data and are therefore more important in the reconstruction of the stream. As P-frames and B-frames contain data relating an I-frame, the loss of an I-frame can result in errors propagating through the stream. I-frames also have the largest payload of the three frame types. Due to this, a stream of only I-frames would have the best quality but would also have the poorest compression. This leads to larger file sizes and greater bandwidth consumption.

I-frame frequency is therefore typically one or two per second to minimise error propagation and file size while maintaining quality. Encoders can dynamically control I-frame frequency based on video content. This feature is particularly important to maintain quality where a scene change has occurred.

The frames that occur from one I-frame to the next are referred to as a Group of Pictures (GOP). The size of the GOP is related to the I-frame frequency and changes accordingly but specific patterns of I, P and B frames must be adhered to. The standard recommends that a GOP is composed of 15 frames (1 I-frame, 4 P-frames and 10 B-frames) however it has been observed that this recommendation is often overlooked [7]. Media players typically have the ability to play out a MPEG-4 video that does not conform to the standard. It may therefore be beneficial to have an increased I-frame frequency when the video content contains a high frequency of scene changes. The standard frame pattern is illustrated in figure 3 below.

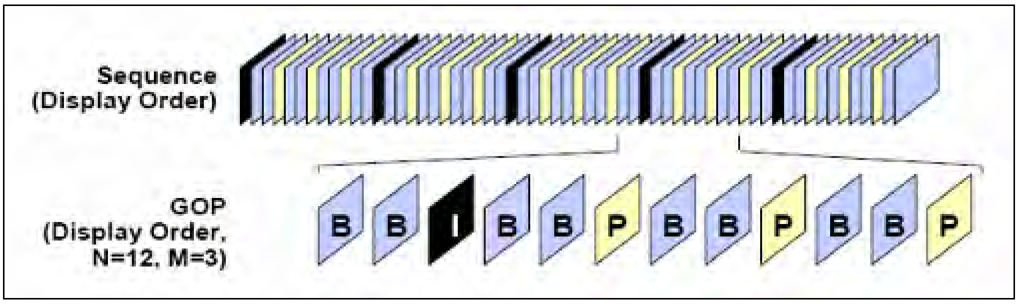

Figure 3: I, P, B frame display sequence.

2.4 Introduction to Wireless Networking

The Institute of Electrical and Electronics Engineers (IEEE) is the most important standardisation body for local area and wireless networks. These standards fall under the IEEE 802.3 and IEEE 802.11 families for Ethernet LANs and WLANs respectively.

The IEEE 802.11 standard was ratified in 1997 and is built upon the evolution of the IEEE 802.3 standard by introducing physical layer (PHY) and medium access control layer (MAC) protocols in order to accommodate wireless communications. It operates in the 2.4GHz unlicensed ISM channel and originally had a maximum throughput of 2Mbps which was deemed to be too slow for most applications [8]. Since then many IEEE 802.11 taskgroups have been initiated which has led to further amendments and enhancements to the IEEE 802.11 standard being ratified. The latest wireless standard, IEEE 802.11n, was ratified in late 2009.

The important wireless standards are:

- IEEE 802.11
- IEEE 802.11b
- IEEE 802.11a
- IEEE 802.11g
- IEEE 802.11e
- IEEE 802.11n
- Several other specific purpose standards

2.5 Wireless Standards

<u>IEEE 802.11b</u>

The IEEE 802.11b standard was ratified in July 1999 and utilises the same unlicensed 2.4 GHz ISM radio channel as the original IEEE 802.11 standard. It supports transmission rates up to 11 Mbps. There are four transmission rates defined in the standard and these are 1 Mbps, 2 Mbps, 5.5 Mbps and 11 Mbps. As it operates in the 2.4 GHz radio channel, IEEE 802.11b devices are susceptible to interference from other appliances like microwave ovens.

IEEE 802.11a

The IEEE 802.11a workgroup was developing this standard at the same time as the IEEE 802.11b standard was being developed. It operates in the 5 GHz unlicensed ISM radio band and supports bandwidths up to 54 Mbps. It is defined for transmission rates of 6 Mbps, 9 Mbps, 12 Mbps, 18 Mbps, 24 Mbps, 36 Mbps, 48 Mbps and 54 Mbps. Due to the higher frequency, the range of IEEE 802.11a is shorter than that of IEEE 802.11b and it is also less capable at penetrating walls and obstructions. There is, however, generally less interference in these bands from industrial sources.

IEEE 802.11g

The IEEE 802.11g standard was ratified in 2003 and supports transmission rates up to 54 Mbps. Unlike IEEE 802.11a it operates in the unlicensed 2.4 GHz ISM band. The IEEE 802.11g standard attempts to combine the increased range of IEEE 802.11b with the bandwidth of IEEE 802.11a. IEEE 802.11g network adapters and devices are designed to be backwards compatible with IEEE 802.11b devices.

IEEE 802.11e

The first draft of the IEEE 802.11e standard was available in late 2001 and was eventually ratified in 2005. It aims to address QoS requirements by defining a MAC layer enhancement to address the delivery of voice and video data over wireless networks. It is backwards compatible with the original MAC mechanisms.

Four ACs are defined and are typically labelled Best Effort (BE), Background (BK), Voice (VO) and Video (VI). Each AC can be assigned different MAC layer parameters in order to prioritise individual traffic streams.

IEEE 802.11n

The IEEE 802.11n standard was ratified in late 2009 and aims to take advantage of Multiple Input Multiple Output (MIMO) technology to significantly increase the available bandwidth to a maximum of 300Mbps. It operates in both the 2.4GHz band and the 5 GHz band.

2.6 Medium Access Mechanisms

The original IEEE 802.11 standard includes the definitions of the MAC and PHY layer. The MAC layer has the ability to utilize one of two access mechanisms; the DCF (Distributed Coordination Function) and the PCF (Point Coordination Function). DCF is based on carrier sense multiple access with collision avoidance (CSMA/CA) technology. It operates on a best effort principle and all stations have equal opportunity to contend for access. The PCF employs a centrally managed polling mechanism to control data transmissions. The use of PCF is optional as stated in the standard and by and large has been ignored by the major equipment manufacturers.

In DCF mode, all stations must sense the medium to be idle before transmitting. When the medium has been idle for a specified period of time known as a distributed interframe space (DIFS) the station sensing the medium begins to transmit. The DIFS for IEEE 802.11b networks is 50μs and 28μs for IEEE 802.11g networks provided that no IEEE 802.11b nodes are present. When only IEEE 802.11g devices are present in a topology the network is said to be operating in “pure G” mode.

If a station has data to transmit and senses the medium to be busy it defers its transmission until the station using the channel has finished its transmission. At this point the deferring station selects a random backoff value between zero and the contention window (CW) and decrements this value while the medium is sensed to be idle. If the backoff counter finishes and the medium has remained idle the station begins its transmission.

An acknowledgement scheme is employed that requires a positive acknowledgement to be received within an interval of SIFS. If an acknowledgement is not received because (a) the packet has been lost, (b) the packet has been corrupted of if (c) the acknowledgement has been lost, the transmission is deemed unsuccessful. In this instance a new backoff procedure commences with a new backoff counter between zero and twice the CW. The CW is an integer value between the minimum CW (*CWmin*) and the maximum CW (*CWmax*). For IEEE 802.11b devices *CWmin* = 31 and *CWmax* = 1023.

Figure 4 [9] illustrates this backoff procedure.

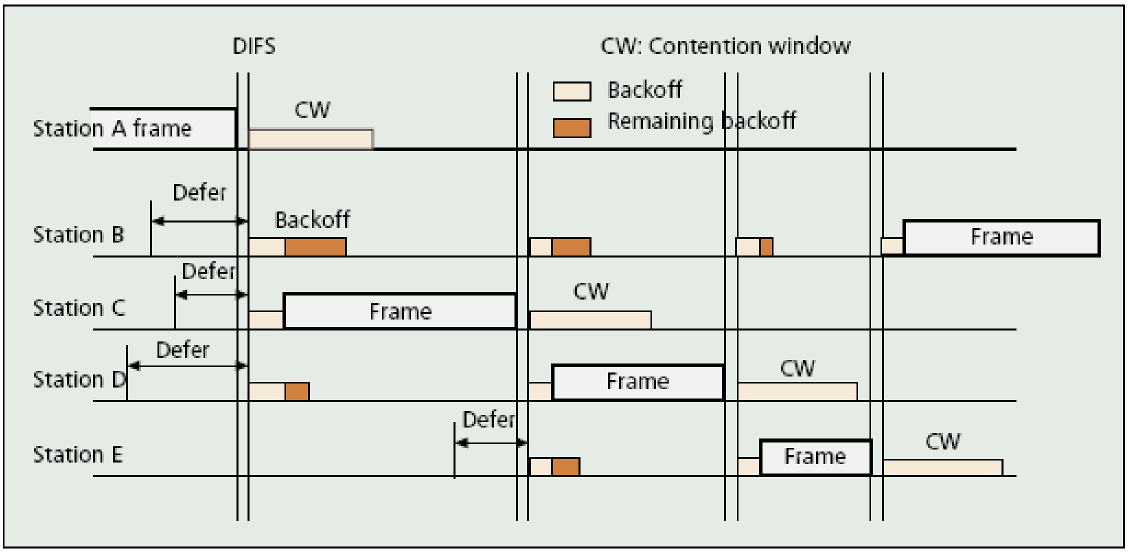

Figure 4: Backoff Procedure

As the DCF selects a random backoff based on the *CWmin* and *CWmax* values as stated in the standard for the PHY layer, all stations are given an equal opportunity to access the medium in order to transmit their data.

The best effort MAC mechanism defined in the original IEEE 802.11 standard does not suit some voice and video applications that have strict requirements for bandwidth, delay, jitter and packet loss. In order to accommodate these applications and to make provisions for Quality of Service (QoS) requirements the IEEE 802.11e standard was developed. It provides for a differentiated service for prioritising data streams by employing a modified DCF. This mechanism, the Enhanced Distributed Channel Access (EDCA) allows four separate traffic streams to be defined and prioritised accordingly. Each traffic category is provisioned with its own data queue in the AP and contends for access with internal collision resolution (Figure 5). Typically, the four access categories are labelled Voice (VO), Video (VI), Background (BK) and Best Effort (BE).

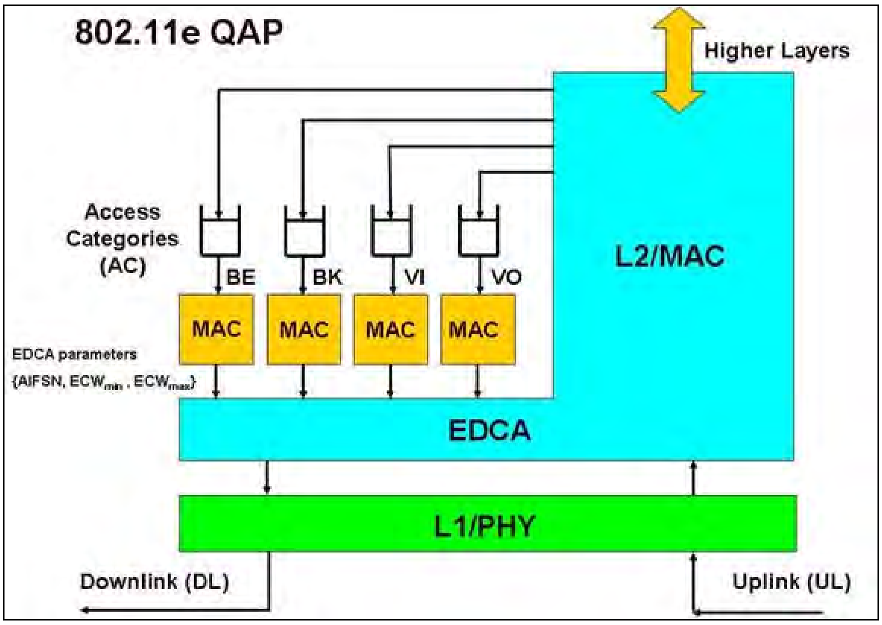

Figure 5: 802.11e Access Categories [9a]

Differentiation is achieved by assigning a transmission opportunity (*TXOP*) value for each AC. This value determines the period of time during which a traffic category can initiate a transmission. This allows the AC to send multiple packets without having to re-contend for access within the *TXOP* duration.

Each traffic category can also have its own values for *CWmin* and *CWmax*. By ensuring that one AC always selects from a smaller range of CW it has a higher probability of winning a transmission opportunity due to its shorter backoff period. IEEE 802.11e also uses an Arbitration Interframe Space (AIFS) instead of *DIFS*. The *AIFS* is always at least equal to *DIFS* and can be increased for each traffic category individually. An exception to this rule allows APs to have an *AIFSN* equal to one. Access Categories with smaller *AIFS* values defer for a shorter space of time than those with higher *AIFS* values allowing for more frequent access to the medium. The relationship between *AIFS* and *AIFSN* is determined by the equation below.

$$AIFS[AC] = AIFSN[AC] \times SlotTime + SIFS \tag{2.1}$$

Figure 6 below illustrated the EDCA access mechanism [9].

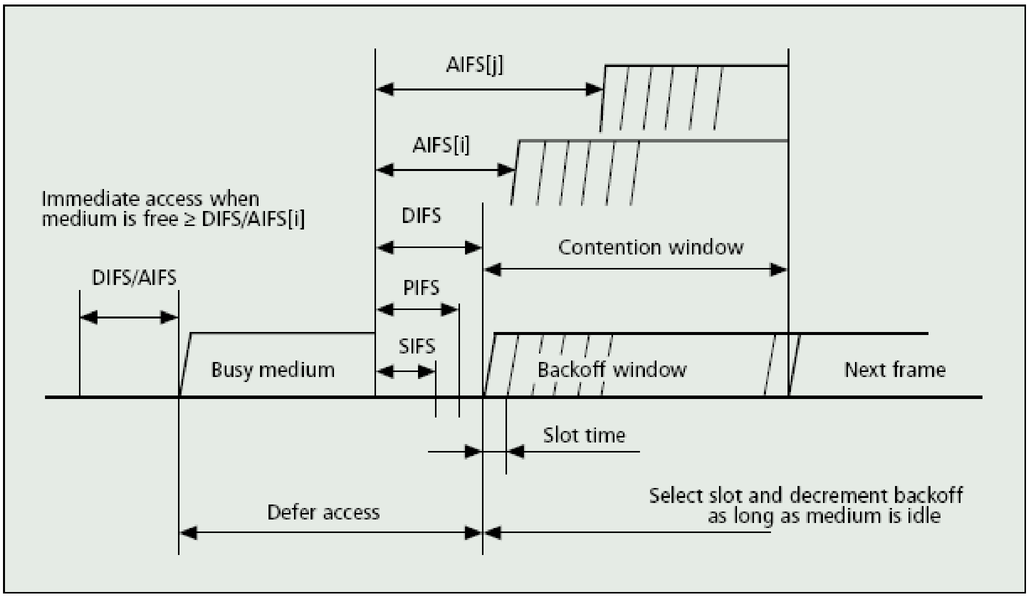

Figure 6: EDCF Access Mechanism [9]

In short, traffic categories that have smaller *AIFSN*, *CWmax* and *CWmin* values and higher *TXOP* values will be more likely to win contention opportunities and have the ability to transmit more than one packet respectively.

2.6 Video Streaming

Streaming is the act of playing out a file on a client station while it is still being actively downloaded from the source or server. It can be applied to video and audio media (i.e. Internet radio and television) as well as other forms of data streams: live data, weather, and sports results.

The amount of bandwidth required by different types of streams can vary dramatically. Video streaming places an onerous requirement for bandwidth on a network based on its frame rate, resolution and bit rate. As these parameters increase, so too does the bandwidth required to transmit the video file.

Video streaming can be offered as either a real time or on-demand service. Real time streaming services are used for video conferencing applications and require low delay times and packet loss due to the presence of end user interaction. In video conferencing applications each end user is both a client and a server of a video stream. The data can be streamed in a unicast session to one other client or in a multicast session to facilitate several other clients.

The medical industry is currently developing systems to facilitate remote medical consultation by employing current streaming and networking technologies.

Dublin's Tallaght hospital currently has a robotic device on trial [10] that allows stroke patients to be monitored and examined by a medical specialist remotely. Developed by Intouch Healthcare in the USA, the RP7 robot allows specialists to interact with patients audibly and visually by employing a range of cameras and microphones. The patient is able to hear and see the specialist by way of speakers and an LCD display on the robot. The RP7 also has the ability to interface with electronic stethoscopes, otoscopes and ultrasound devices via an expansion port allowing the specialist to take readings remotely. As the robot is motorised and can move around the hospital under the specialists control it cannot be tethered via wires to power outlets or communications ports. All of the features of the device are controlled via a control station and joystick and audiovisual data and test results are transmitted using IEEE 802.11 and broadband networks.

On-Demand services provide pre-recorded and preformatted material for streaming. The content is stored on a server and play out is initiated when a user request for content is received. Many popular video hosting websites and services are based on this principle, e.g. Youtube, Vimeo, Google Video. This service differs from real time services as the content is usually available at all times, real time services are only available as they happen. Increasingly TV stations (BBC, Channel 4, RTÉ) are offering VoD or "playback" services for their customers.

2.8 Video Quality of Service

QoS is the term given to describe the overall quality of a video when it has reached the client side play-out device. It is composed of two separate elements; Quality of Delivery (QoD) and Quality of Experience (QoE).

Quality of Delivery relates to how a stream is affected by network characteristics such as jitter, delay, packet loss and throughput. These characteristics can be easily quantified at the network layer using conventional tools.

Quality of Experience relates to how an end user perceives the quality of the played out video. The end user's perception is determined by the human visual system (HVS). The HVS is an extremely complex system with many features including spatial frequency sensitivity, luminance masking, texture masking, temporal frequency sensitivity and short term memory effect. A person's perception can also be modified using external stimulants such as alcohol, coffee and nicotine and their physical condition, including tiredness, can also be an influencing factor affecting perception. A user's QoE can also be influenced by their personal experience and expectation of the service. QoE is a more difficult area to quantify and numerous metrics and techniques have been devised. The problem in quantifying QoE arises as end users can perceive the same video, transmitted under the same network conditions, in a variety of ways. QoE can be subjectively tested using human live trials and the results statistically analysed but these trials are time consuming and costly to implement.

Objective metrics have been developed with the intention of estimating end-user quality of experience. Among these is the Peak Signal to Noise Ratio (PSNR). This metric requires both the host and the client side video files to be available for test. It compares the host and the client side videos on a pixel by pixel and a frame by frame basis, and returns a decibel (dB) value for the entire video clip. It has been found that values between 20dB and 25dB [11][12] are acceptable for video transmission over wireless networks.

This metric is simple to calculate but has some limitations. For instance, as it operates on a frame by frame basis and returns a single value for each frame, it cannot distinguish between a catastrophic failure in a small section of an image and a smaller discrepancy applied over an entire image. Both of these errors may yield the same PSNR value although either one may be more acceptable to an end user.

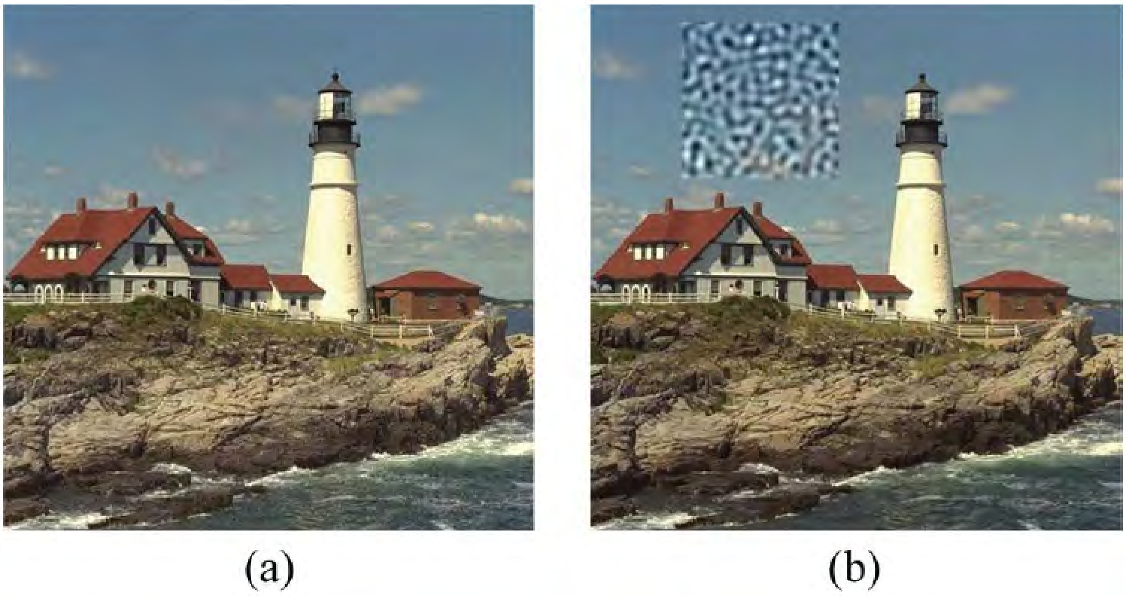

Figure 7: Examples of corrupt video frames yielding the same PSNR

Figure 7 above shows two images that yield the same PSNR value. Image (a) has high frequency noise added almost exclusively to the bottom region of the image where there is already high visual complexity caused by the water, shoreline, textures and edges. This serves to mask the noise from human vision. Image (b) has low frequency noise added to a more localised area of smooth blue sky in a more noticeable pattern.

2.8.1 Quality Metrics

In order to determine the end user QoE of a streamed video a quality metric is required. Where live trials are employed the metric used is a Mean Opinion Score (MOS). Generally observers are asked to rate a clip from 1 to 5 with 1 being the poorest quality and 5 being the highest quality. These values are then statistically analysed to obtain a MOS for the video. Live trials are time consuming and expensive to conduct, they also present difficulties as the viewing environment must be strictly controlled throughout the duration of the experiment.

In order to overcome these difficulties objective metrics have been developed to obtain quality scores without the need for subjective testing. Although it is well recognised that live trials yield the most accurate results, objective testing is far more practical and flexible. The test of an objective metric is how closely it can estimate what the MOS of a video clip would be. Objective metrics fall into three separate categories: Full Reference, No Reference and Reduced Reference.

Full reference (FR) metrics require both the host side and client side videos to be available and are therefore most suited to offline video and image testing. They perform a frame by frame comparison of the two video files to yield their result. The full reference PSNR metric cannot be used for MPEG-4 video as it does not contain data for every pixel of every frame. MPEG-4 videos must be re-encoded into a suitable format first. For the purposes of the work described in this thesis the YUV format has been utilised. This format contains data for each pixel of each frame of a video file.

No Reference (NR) metrics analyse only the client side video file to make their estimation. This makes them more flexible than FR metrics as the host side video file is not always available. NR metrics must be able to distinguish between image content and image distortion requiring complex processing in order to give accurate results.

Reduced Reference (RR) metrics fall between FR and NR metrics. They extract a number of features from both the host and client side videos and make their analysis on these features alone, typically motion and spatial details are used. RR metrics require a separate communications channel back to the host side video in order to obtain information relating to it.

As both the host and client side video files are available, full reference metrics have been employed for the analysis detailed in this thesis.

2.9 Challenges Associated with Video Streaming over Wireless Networks

Research into improving video services over the wireless medium is an active topic with areas such as encoding, content complexity, buffer management, streaming servers, compression techniques, adaptation and physical layer attributes all receiving attention from many institutes and researchers.

Due to the large file sizes associated with digital video files, long periods of downloading before viewing is an unacceptable system. Downloading while playing, or streaming, is the obvious solution but in order to work satisfactorily it is imperative that each frame arrives on or before its play out time. This task is made all the more complicated when the video stream is transmitted over the wireless medium.

IEEE 802.11a/b/g WLANs also use lower transmission rates compared to traditional wired LANs. IEEE 802.11a/g offers a maximum data rate of 54Mbps and IEEE 802.11b offers a maximum of 11Mbps. In reality the achievable maximum data rates for these networks is significantly lower than the theoretical rate due to the access method and associated protocol overheads. IEEE 802.11g also employs “protection mechanisms” to avoid interfering with legacy IEEE 802.11b devices – these can often result in a throughput performance that is poorer than that for IEEE 802.11b [13].

WLANs also suffer from high error rates caused by fading, interference, reflections and changes in the propagation environment adding to the challenge of providing acceptable video QoE.

The video stream itself can present its own set of challenges for video streaming. Content providers have many parameters to choose from such as codecs, encoding policies, resolutions and frame rates when preparing a video for streaming and each can present varying bandwidths, packet sizes and frequencies.

2.10 Chapter Summary

Video streaming presents a significant challenge in both execution and analysis. Many configuration settings exist for any single video transmission which further increases the complexities involved.

The limited bandwidth, contention mechanism and hostile nature of the wireless medium present many of the most significant challenges for video streaming. The newly ratified IEEE 802.11n protocol increases the maximum transmission rate of the wireless medium up to 300Mbps by employing multiple antennas and Multiple Input Multiple Output (MIMO) technology. However, it can be assumed that as the bandwidth available to end users increases their demand also increases leading to similar problems in the future.

The IEEE 802.11e protocol provides a mechanism to prioritise traffic streams in order to overcome the limitations of the access mechanism. Unfortunately this mechanism presents a large number of variables to network operators which are usually only set once. In order to take full advantage of the IEEE 802.11e mechanism, the EDCA values need to change with the characteristics of the network and the demands placed upon it.

Determining the quality of a received video stream is not a straightforward task. Although it is widely accepted that live trials yield the most accurate results, they are difficult to perform in practice as they require considerable resources in terms of people and time. To compensate for this, several quality metrics have been developed which attempt to estimate the results that would be obtained from live trials. Of these the PSNR is widely recognised and utilised although it does have the limitation that all impairments are weighted equally. This thesis proposes to determine the optimal values for *Cmin* to supply to the CNRI WRRC RRM application in order to provide high quality video over dynamically tuned IEEE 802.11e WLANs.

CHAPTER 3 LITERATURE REVIEW

Video streaming over the wireless medium is an active research topic with a considerable body of work covering the areas relating to coding/encoding [14], QoS provisioning [15], admission control [16], QoE evaluation [17] and adaptive streaming techniques [18]. This thesis is primarily concerned with QoS provisioning for video streaming applications on WLANs through the use of the IEEE 802.11e mechanism. Previous works in these areas are described in this section.

3.1 Video Transmission

Ksentini, Gueroui and Naimi [19] have proposed a system that improves user QoE by splitting different video layers of H.264 video files into different IEEE 802.11e ACs. The EDCA values and retry limits for the ACs were statically set and the values used are shown below in Table 2.

	AIFSN (µs)	CWmin	*CWmax*	Queue Length	Max Retry Limit
Parameter Set Information (AC3)	2	7	15	50	8
IDR and Partition A (AC2)	2	15	31	50	8
Partition B and C (AC1)	2	31	1023	50	4
Background Traffics (AC0)	3	31	1023	50	4

Table 2: QoS values used by Ksentini, Gueroui and Naimi

Through NS-2 simulation a H.264 video file is unicast from server to client while four other wireless stations contend for access by generating 300 kbps CBR background traffic streams. Their research found that there was no increase in loss for IDR frames (AC2) compared to EDCA and DCF approaches (Table 3 and Table 4 respectively) when the background traffic was increased. In the case of AC1, their approach had a higher loss rate compared to EDCA and DCF approaches due to the lower priority and smaller max retry limit of the AC.

	AIFSN	CWmin	*CWmax*	Queue Length
H.264 Streams (AC2)	2	15	31	50
Background Traffic (AC0)	3	31	1023	50

Table 3: EDCA values used. (Ksentini et al.)

DIFS (µs)	CW	Queue Length
30	31	50

Table 4: DCF parameters. (Ksentini et al.)

They also found that the lowest priority AC (AC0) where background traffic was transmitted had an increased loss rate when compared to the standard EDCA and DCF settings. In this instance the video has experienced a reduced loss rate at the expense of the lowest priority AC, containing the background traffic, experiencing an increased loss rate. Overall their approach yielded an average increase of 15dB in QoE as the more important ACs were given higher priority on the medium. Utilising this approach there was also a decrease in the amount of frames that could not be decoded at the client side when compared to EDCA and DCF approaches.

Shin and Chung [7] proposed a cross layer rate control scheme for MPEG-4 transmission over an IEEE 802.11e network. Their system places the different MPEG-4 frame types (I, P, and B) into the different IEEE 802.11e ACs. The EDCA values for each AC were statically set and are shown in Table 4 below. Although these values were chosen to prioritise the ACs, it is unclear how these particular values were chosen.

Type	**AIFS (μs)**	**CWmin**	***CWmax***	**Max retry limit**
I-Frame (AC3)	50	7	15	8
P-Frame (AC2)	50	15	31	8
B-Frame (AC1)	50	31	1023	4
Best Effort (AC0)	70	31	1023	4

Table 5: Parameters of the 802.11e ACs (Shin and Chung)

Information from a feed back loop from the network is used to drop frames according to the current estimated network bandwidth and frame priority. In order to estimate the available network bandwidth a new algorithm was designed that takes into account the bandwidth available in each AC and estimates the total available bandwidth based on these values. This system was simulated using NS-2 with a 500 kbps video traffic stream and a background load of 700 kbps. They showed that their AC aware bandwidth estimation algorithm yielded more accurate bandwidth estimation results than an AC ignorant estimation. They also found their system led to a decrease in the packet loss rate resulting in an improvement in QoS.

Tu and Sreenan [20] presented an adaptive split transmission system for video streaming in Wireless Mesh Networks (WMNs). Their system utilizes multi-radio devices and endeavours to use the free bandwidth on several channels to transmit video data if it is deemed that the video stream will overload a single channel. Two scenarios have been simulated using NS-2; the first scenario simulates a single sender and receiver. Each has four radio interfaces with one channel per interface. Channel bandwidth is set at 2 Mbps. One channel is used as a control channel and three are used as data channels. The video traffic load is set to 128 kbps. Under “good” conditions the video is transmitted from the sender to the receiver through channel 1. As background traffic is increased the algorithm splits the video data into multiple streams and uses other channels. They found that with a single sender/receiver 5100 kbps of video traffic could be transmitted that met delay constraints compared to 1500 kbps without employing the algorithm.

This system was also tested with multiple receivers where 25 nodes with six radio interfaces were simulated. Each interface has one channel with a set bandwidth of 2 Mbps, two channels were used as control channels and four for data. Four video flows are sent from one sender to four randomly selected receivers. Channel 1 is used for all streams until it becomes overloaded, at which point the adaptive split transmission algorithm is employed. Under these conditions 850 kbps of video traffic could be sent that met the delay constraints compared to only 150 kbps that could be sent without employing the algorithm.

MacKenzie, Hands and O’Farrell [21] employed a similar technique to Ksentini et al. In their work, the different video layers of a H.264 video were transmitted through different IEEE 802.11e ACs according to Table 6 below.

Scheme	**PSI**	**I-Slices**	**P-Slices**	**B-Slices**
Default	AC_BE	AC_BE	AC_BE	AC_BE
Scheme 1	AC_VI	AC_VI	AC_VI	AC_VI
Scheme 2	AC_VO	AC_VI	AC_VI	AC_BE
Scheme 3	AC_VO	AC_VO	AC_VI	AC_BE

Table 6: Mapping of video layers to ACs for four different schemes

The EDCA values used for each AC were set to their recommended default values as shown below in Table 7

AC	*CWmin*	*CWmax*	*AIFSN*	*TXOP_limit*
AC_VO	7	15	2	3.264ms
AC_VI	15	31	2	6.016ms
AC_BE	31	1023	3	0
AC_BK	31	1023	7	0

Table 7: EDCA values for each AC (MacKensie et al.)

The number of concurrent streams that each scheme could accommodate was then evaluated. This work was simulated using the NS-2 simulation software. Three different video clips were chosen based on their visual complexity and were encoded at 2 Mbps and 4 Mbps with a resolution of 720x576 pixels. They found that the lowest overall packet loss rate was achieved when scheme 1 was employed. Under subjective testing it was found that schemes 2 and 3 yielded more acceptable MOS values than scheme 1 as the number of concurrent video streams was increased. This result shows that packet loss (QoD) is not directly related to MOS (i.e. QoE).

3.2 Quality Evaluation

Evaluating the quality of a received video stream is not a trivial task. Subjective testing represents the most accurate method for obtaining quality ratings [22] but live-trials are difficult, time consuming and costly to perform. To solve this problem objective testing techniques have been developed and advanced. As the HVS is difficult to model, developing objective metrics is a challenging task. There have been two main approaches to the development of objective testing. The vision modelling approach aims to model aspects of the HVS to predict the perceived quality.

The engineering approach looks for artefacts, blockiness, blur and jerkyness in a video and combines them to predict the overall quality.

Gao et al. [23] examined how QoE of several different video clips, encoded at different bitrates, would be affected by changes in *AIFSN* and *CWmin*. In their work they varied the *CWmin* value from 10 to 60 in steps of 5 while the *AIFSN* remained constant and performed a second set of experiments where the *AIFSN* was varied from 4 to 21 in steps of 1 while the value of *CWmin* remained fixed. Tables of the EDCA values for both experiments are shown below. (Table 8 and Table 9)

Type	**Priority**	**CWmin**	***CWmax***	***AIFSN***
Voice	High	7	1023	3
Video	Medium	10 ~ 60	1023	4
Data	Low	63	1023	5

Table 8: EDCA settings used for varying *CWmin* experiment

Type	**Priority**	**CWmin**	***CWmax***	***AIFSN***
Voice	High	7	1023	3
Video	Medium	15	1023	4 ~ 21
Data	Low	31	1023	22

Table 9: EDCA settings used for varying *AIFSN* experiment

The values for *CWmin* and *AIFSN* chosen for this experiment are of some concern as *CWmin* and *AIFSN* values are usually in the range

$$CW_{\min} = 2^{x} - 1, \text{ where } 1 \leq x \leq 10 \tag{3.1}$$

$$\text{and } 1 \leq AIFSN \leq 15 \tag{3.2}$$

The experiments were simulated using NS-2 for a single QoS enabled AP (QAP) and 15 wireless STAs. Each station transmitted mixed format data according to Table 10 below.

Type	Inter-arrival Time (msec)	Packet Size (bytes)	Data Rate
Voice	20	160	64kbps
Video	10	1280	1Mbps
CBR	12.5	200	125kbps
FTP	-	1000	-

Table 10: Traffic types and characteristics used (Gao et al.)

In the simulation one of the stations transmits video data according to a trace file previously captured using the TCPDump software.

A client side video is then generated by degrading the host side video according to packet loss data provided by the NS-2 program. The host and client side videos are then objectively compared and expressed as VQM values. Please refer to section 4.3.2 for further information on the VQM metric.

They concluded that different video clips transmitted under the same network conditions can yield dissimilar video qualities. Gao et al. observed that two different clips, transmitted under the same conditions, that suffered similar packet loss rates produced video quality values that varied dramatically. They also concluded that a more intelligent network control method to improve end-user satisfaction based on network conditions and video content would be more appropriate.

Wang and Bovik [24] present an objective metric that aims to incorporate aspects of the HVS; namely frequency sensitivity, luminance masking, texture masking, temporal frequency sensitivity and short-term memory effect. Their system compares the original frame and the test frame and calculates the mean square error (MSE) to use as an initial error map. The error map is then weighted according to further tests to model the aspects of the HVS mentioned above.

Pinson and Wolf [25] proposed a reduced-reference general model VQM to provide estimates of the overall impression of video quality. Their system requires an ancillary data channel with a bandwidth of 14% of the uncompressed video stream. Their metric was compared to 17 other quality metrics and statistically outperformed 16 of those in a 525 horizontal line test when compared to subjective results. In a 625 line test the VQM was in the top 4 of the 18 models. As a result the VQM was standardised by the Institute for Telecommunication Sciences (ITS) in July 2003.

Further objective metrics were developed by Wang, Bovik, Sheikh and Simoncelli [26]. In this paper, a new metric, the Structural Similarity (SSIM) measurement system is proposed. This work is based previous work undertaken by Wang [27], [28]. The SSIM is based on the hypothesis that the HVS is highly adapted to extracting structural information. Subjective testing was undertaken to evaluate the Mean Opinion Score (MOS) for an array of images. The images were all compressed using the JPEG and

JPEG2000 standards and varying quality levels. The original database composed of 29 high resolution images which yielded 175 JPEG images and 169 JPEG2000 images.
They found that if the PSNR metric is used as the benchmark for MOS prediction, their model outperformed the PSNR, UQI [29] and Sarnoff quality metric models and gave more realistic MOS predictions.

Lin and Chau [30] expanded upon the work detailed in [26] and introduced the hierarchical SSIM metric. Their argument is that the Mean SSIM (MSSIM) of an image underestimates the magnitude of annoyance caused by blocking and artifacting. Their system divides each image into *M* by *N* blocks. The MSSIM is calculated for each block and also for the whole image. The hierarchical MSSIM is then computed as a function of global MSSIM and the minimum value of MSSIM for a single block of the image.

$$hierMSSIM = [mnMSSIM]^{\alpha} \times [miMSSIM]^{\beta} \qquad (3.3)$$

where *mnMSSIM* is the global MSSIM of the image and *miMSSIM* is the lowest value of MSSIM for a single individual block of an image.
They tested their system on a range of JPEG and JPEG2000 images that emulate typical errors seen in MPEG/H.26x video streams.
They conclude that if the measure of a metric is how closely it correlates to human perception then this metric outperforms both the PSNR and SSIM metrics.

3.3 State of the Art

Network aware encoding or adaptive encoding [31] has the ability to vary the bitrate of the video stream based on the current network conditions. If there is an increase in the bandwidth consumption the encoder can decrease the bitrate of the video and thus decrease the bandwidth requirement of the stream. The reason for developing this system is that a low quality stream is more acceptable to the end user than the complete failure of a higher quality stream. This is an adaptive system that reduces the quality of the video if the bandwidth available is reduced, it could be argued that it would be more logical to preserve the bandwidth allocated to the video stream and to punish the other non real time traffic streams where retransmissions are possible. The way in which the bitrate is usually modified is based upon QoD parameters which do not correlate to end user QoE.

Ciubotaru, Muntean and Ghinea [18] introduce a Region of Interest based Adaptive Scheme (ROIAS). Their system aims to adaptively tune the video bitrate based on current network conditions and adjusts regions of each frame based on the user interest in the region. Their approach utilises the bandwidth available to maintain the quality in the most important areas of the frame. Their work presents PSNR and VQM differences for the entire frame and the area of maximum user interest (AMUI) for 2 versions of their algorithm. The highest gain in PSNR for the AMUI recorded is 28%, however the PSNR for the entire image was reduced by 25% when compared to a quality oriented adaptation scheme [32].

Split streaming [6], [18] is present in many forms. The more widely accepted version of adaptive split streaming places the different frames of a video stream, I, P and B in the case of MPEG-4, into different access categories of the IEEE 802.11e mechanism. Using this system the I-frames can be placed in the highest priority AC with the P-frames and B-frames in lower priority ACs. The work detailed has shown reductions in packet loss and an increase in QoE under simulated conditions. The rationale behind this system is that the P-frames and B-frames are of little use without the reference I-frame therefore the I-frame should be given the best chance possible to be transmitted.

3.4 Chapter Summary

It is evident that there is a large body of published works in the area regarding video streaming over wireless networks and QoS provisioning. In this chapter several reports have been discussed where the IEEE 802.11e protocol has been utilised in order to increase video quality. In the literature discussed above, two main limitations consistently occur.

In these papers the results presented are based on computer simulations carried out using the NS-2 simulator program. In several cases streaming servers have been used to stream a video between host and client. This stream has then been captured using the TCPdump software to generate a trace file of the streamed video. This trace file has then been used as the input to the simulation package. While simulation results are acceptable there is a need for these results to be experimentally validated. In the cases where the IEEE 802.11e protocol has been utilised there is little or no explanation provided as to why particular EDCA settings have been chosen. These values have also been statically set for the duration of the simulation. In order to take full advantage of the IEEE 802.11e mechanism the EDCA settings need to change according to the conditions on the network.

When a simulation package is being employed to generate results it is quite easy to simulate scenarios that would be extremely difficult to implement on a real time experimental testbed. Current research is concentrating on network aware encoding and split stream adaptation. Both systems require that the server has access to information about the transmission medium and the end user quality respectively.

CHAPTER 4 EXPERIMENTAL PROCEDURE

4.1 Scope of This Thesis

The work detailed in this thesis is concerned with implementing an RRM scheme on IEEE 802.11e based WLANs to deliver QoS for video streaming applications. The CNRI WLAN Radio Resource Controller (WRRC) has the ability to communicate with a Cisco Aironet 1200 series QAP and to adaptively adjust the EDCA settings based on current network conditions [33]. To evaluate the effect of dynamically controlling the *AIFSN* and *CWmin* parameters based on current network conditions each experiment was initially conducted with static IEEE 802.11e parameters (Table 8).

Protocol	Access Category	*AIFSN*	*CWmin*	*CWmax*	*TXOP* [µs]
Static IEEE 802.11e	**All**	2	5	10	3008
Default IEEE 802.11e	**Background**	7	31	1023	0
	Best Effort	3	31	1023	0
	Video	2	15	31	6016
	Voice	2	7	15	3264

Table 8: Static 802.11e and Default 802.11e settings

These results could then be compared with those obtained while the dynamically tuned IEEE 802.11e protocol was employed.

Although the IEEE 802.11e protocol cannot generate more bandwidth, it is believed that real-time data stream prioritisation and RRM will lead to an increase in end user QoE for the video streams through a greater allocation of the available bandwidth to the video AC..

The CNRI WRRC reduces the number of parameters that need to be set for the IEEE 802.11e protocol to one, namely the *Cmin* value, for each AC. In order for the application to dynamically tune the EDCA settings optimally there is a need for the *Cmin* value to be appropriately chosen. This work aims to determine the optimal value of *Cmin* to use in order to deliver the most desirable end-user quality to the video application.

4.2 Experimental Testbed and Tools

The experimental test bed comprised Dell Optiplex GX280 PCs running the Microsoft Windows XP service pack 2 operating system. These machines are equipped with Intel Pentium 4 2.8GHz processors and have a 512MB RAM. The video server was running the Microsoft Windows Server 2003 service pack 2 OS and is equipped with an Intel Pentium D 2.8GHz processor and 512MB of RAM. In the experiments where multiple clients are involved, subsequent PCs fitted with Intel Celeron 3.06GHz processors and 512MB of RAM were utilised.

The video server PC and the Background traffic generator PC were connected to a Cisco Aironet 1200 series QAP via a Linksys 24-port Ethernet switch. All wireless client PCs were fitted with a Netgear WAG 511 Wireless PCI card and associated with the Cisco 1200 series QAP. The monitoring/controlling PC was fitted with a CACE Technologies AirPcap wireless network adapter [34] to monitor the wireless network activity. This device passively captures (i.e. sniffs) wireless IEEE 802.11 frames on the medium and therefore does not influence the network operation as it does not introduce any additional packets onto the network.

Video test files were generated using the Xilisoft DVD ripper software [34] which allowed for multiple resolutions and bit rates to be used. A hint track was generated for each video file using the mp4creator software [36]. The hint track is utilized by the streaming server software to determine how it should stream the video. In this case the server is instructed as to what the maximum allowed packet size should be. The server then generates the packet stream based on this information. The maximum allowable packet size on a network is referred to as the Maximum Transmission Unit (MTU).

For these tests the Apple Darwin Streaming Server (DSS) software [37] was employed to stream the video files. This is an open source version of the commercial Quicktime Streaming Server (QTSS) but it has some file type limitations compared to the QTSS software. For example the QTSS has the ability to stream Apple Quicktime format streams, but this feature is disabled in the Darwin Streaming Server. The limitations placed on the DSS have no impact on the result gathered here.

Background traffic for these experiments was generated using the Distributed Internet Traffic Generator (D-ITG) [38]. This software allows packet size, packet rate, CoS and packet distribution shape, e.g. CBR, VBR, Poisson, exponential, to be controlled.

At the video client the open source VLC player software developed by VideoLAN [39] was used to play out and record the video streams. The ability to record a streamed video was needed for comparison with the original source video file.

The captured video stream and the host side video files were then converted from MPEG-4 format to YUV format. This was achieved using the ffmpeg software [40]. To convert one file from MPEG-4 to YUV format takes a considerable amount of time (up to four times the duration of the video file being converted). To automate this process a PERL script was employed to batch convert from MPEG-4 to YUV format overnight.

When conversion was completed, the source YUV files and the client side YUV files were compared using the Sunray Image YUVAnalyser software [41]. This software returned the PSNR value for each frame of the clip and also the average PSNR value for the entire clip contained in a Microsoft Excel spreadsheet.

Further analysis of these values was accomplished using both PERL and Matlab scripts [AP1] [AP2] to calculate VQM values and to perform a statistical analysis of the PSNR and VQM values.

Along with the AirPcap adapter the CNRI WRRC software application was used to monitor and control the EDCA parameters of the Cisco IEEE 802.11e enabled AP (QAP). This allowed for the recording of available capacity, bandwidth and individual AC loads and capacities. In control mode it uses a wired feedback loop to the QAP to change the values of *AIFSN* and *CWmin* for each AC.

The WRRC also has a file playback facility which allows for the replaying of a captured trace file. This feature is particularly beneficial to understanding how a network is reacting to the traffic loads placed upon it.

The video flow process is graphically outlined below (Figure 8).

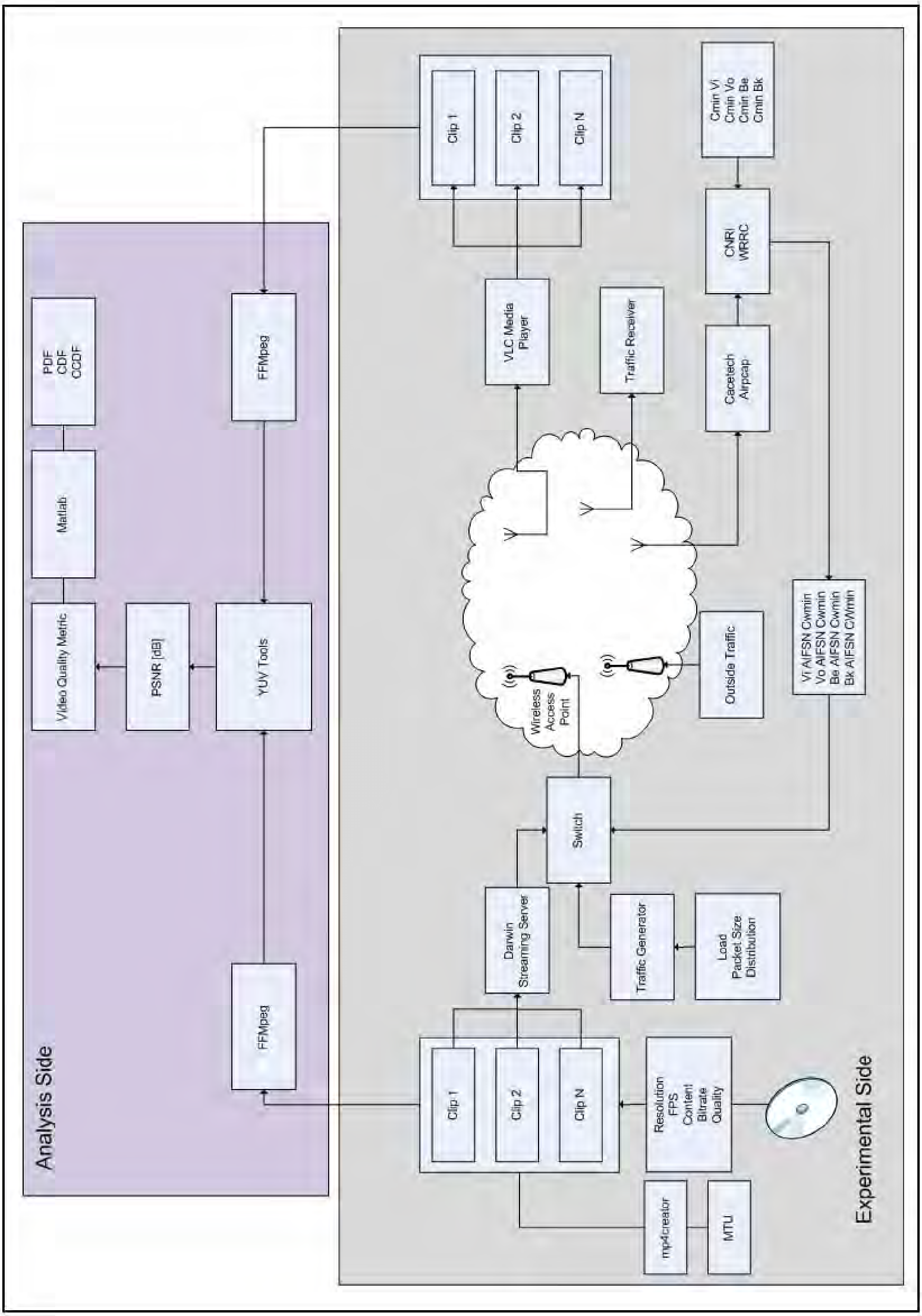

Figure 8: Video Flow Process

4.2.0 Cisco QAP Interface

The Cisco QAP provides the network administrator with several options to communicate with it. A web-based configuration is available by providing the IP address of the QAP to a web browser and providing the username and password when prompted. This option is quite useful as it allows the administrator to easily monitor and make changes to the QAP parameters.

Using this system, a policy has been created on the QAP which maps the packets generated by Darwin Streaming Server to the Video AC and also maps the packets generated by D-ITG to the Best Effort AC based on the DSCP values contained within the header. The figure below illustrates how this is achieved (Figure 9).

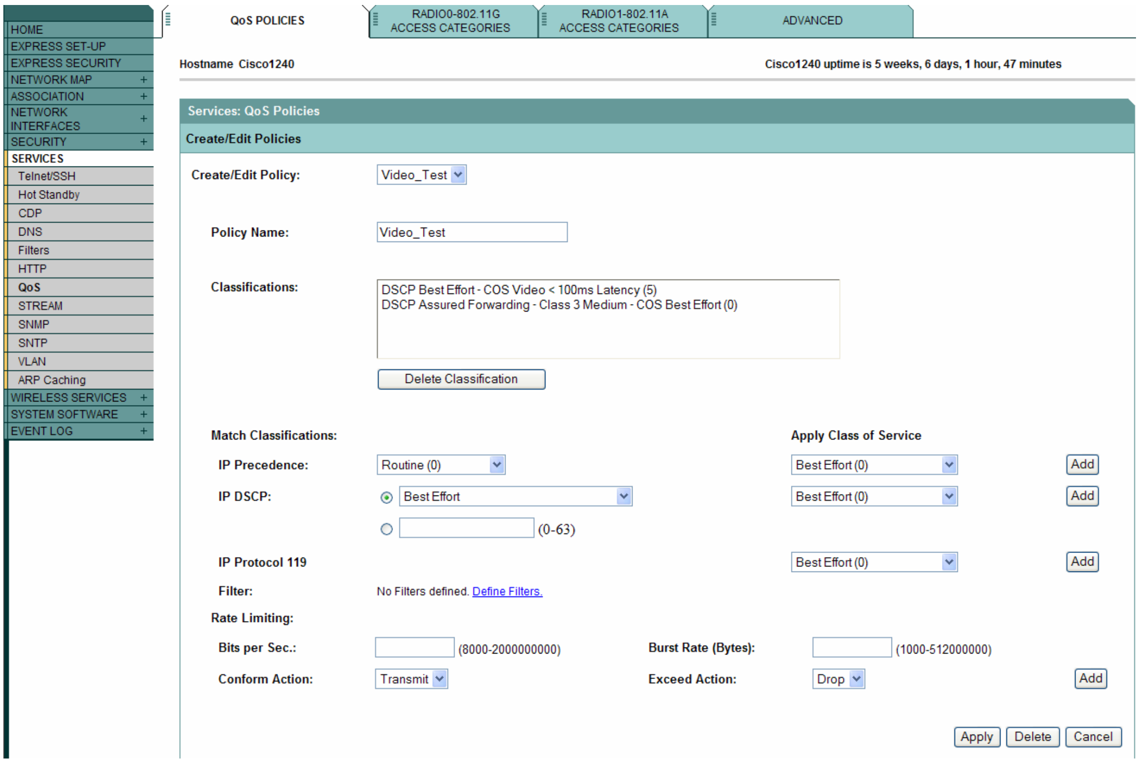

Figure 9: Cisco QAP policy map.

The web-based configuration also allows the EDCA settings to be modified. The figure below demonstrates how these settings can be modified. (Figure 10)

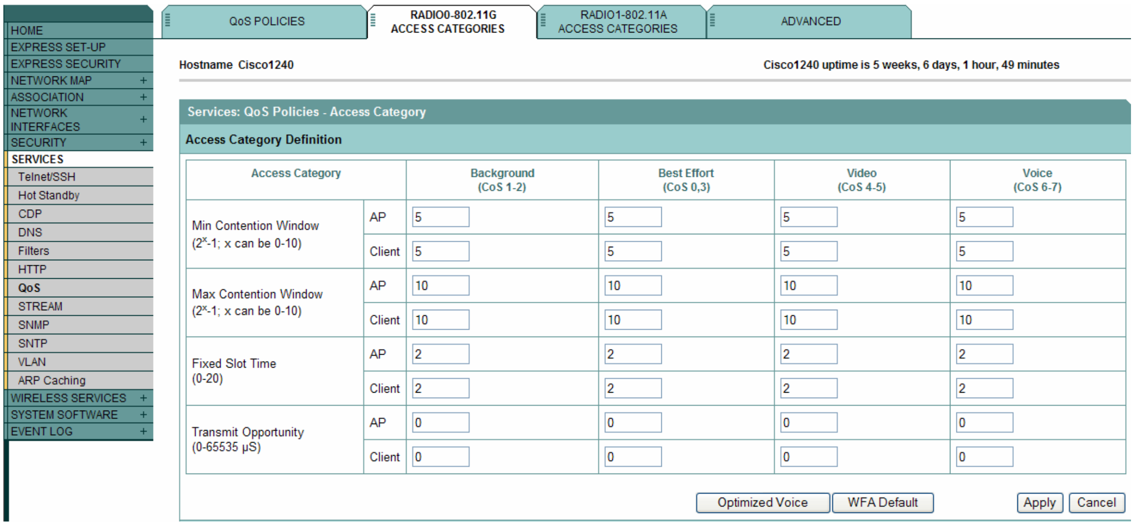

Figure 10: Cisco QAP EDCA settings

The QAP can also be communicated with via a telnet session. This option has the advantage that telnet code can be embedded into a script or program to change the QAP settings. The CNRI WRRC communicates with the Cisco QAP via a telnet session and changes the EDCA settings according to the current network conditions.

4.2.1 CNRI WRRC Application

The WLAN Radio Resource Controller (WRRC) application greatly simplifies the use of the IEEE 802.11e EDCA functionality in implementing radio resource management (RRM) in WLANs. The WRRC automatically tunes the EDCA parameters of a QAP based on a user specified minimum capacity, *Cmin*, for each Access Category (AC) and the current network load.

The operation of the WRRC (Figure 11) is based upon modifying the EDCA settings in order to meet a target Access Efficiency Factor (*AEF*) [33]. The *AEF* is a measure of how efficiently a client or AC can access the medium. Target *AEF* values are calculated for each AC based on the current network conditions and the *Cmin* values specified. The *Cmin* value corresponds to the proportion of the bandwidth that would be available to the station or AC if all other stations or ACs on the network were to increase their load to saturation, therefore, *Cmin* is the minimum bandwidth that is always available to a station or AC.

If the actual *AEF* value is outside a tolerable range of the target *AEF* the *CWmin* and the *AIFSN* values are modified to increase or decrease the actual *AEF* value as necessary.

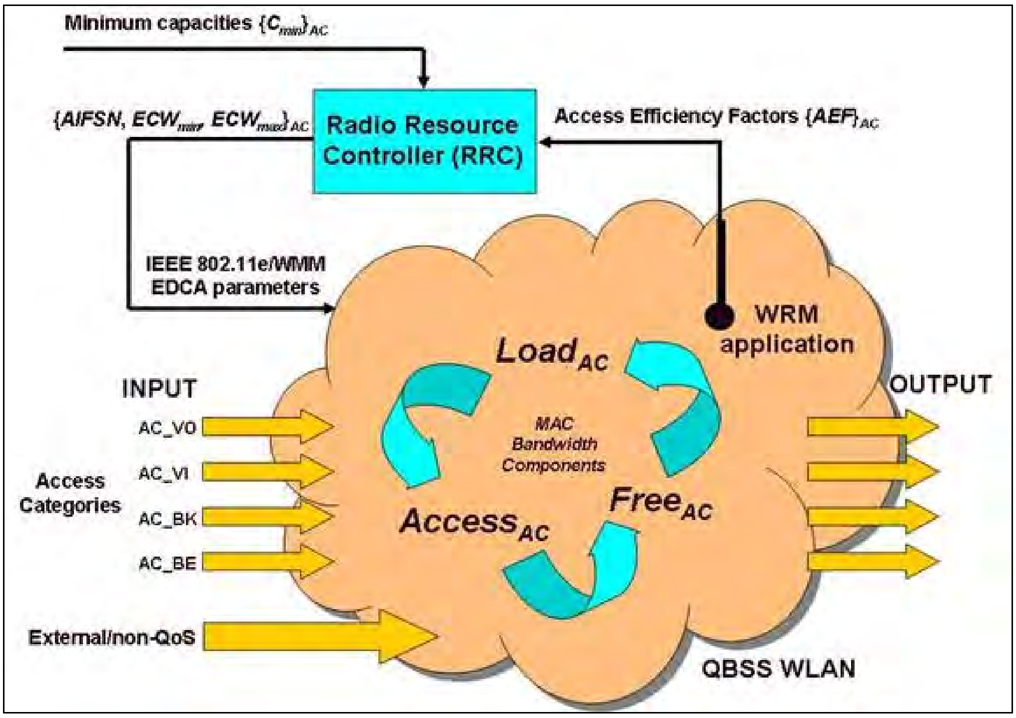

Figure 11: WRRC Operation

On initial startup, the user is presented with a configuration settings screen. This screen allows for the selection of live capture or the playback of a previously recorded trace file. For live capture the source adapter and channel number to monitor are set here. For file playback a file selection field is available. Lastly, for live capture the measurement interval can be specified.

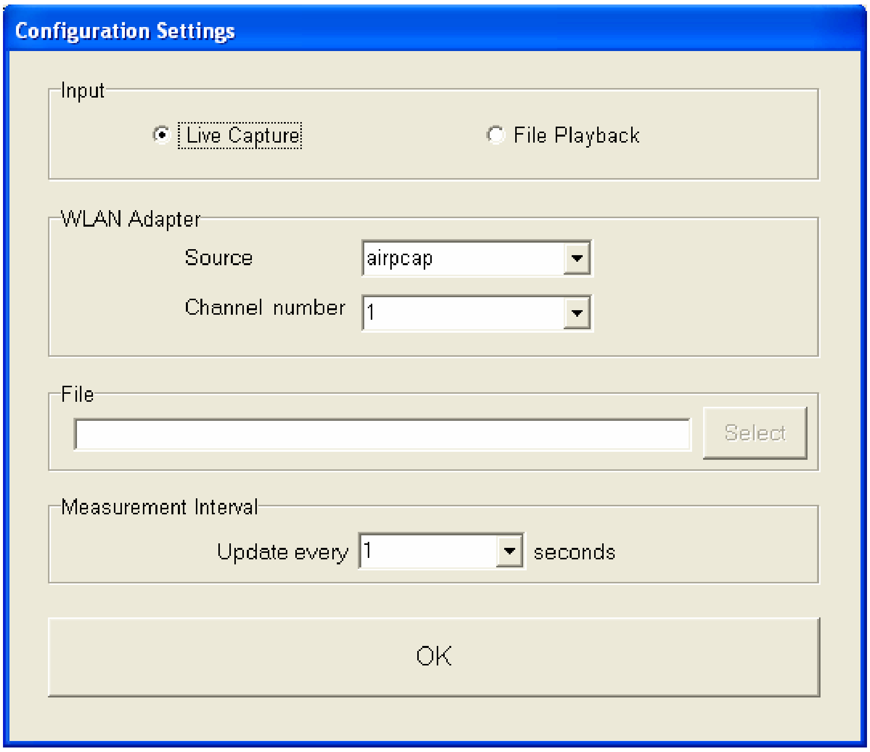

Figure 12: WRRC Configuration Settings Window

A second configuration screen is presented if live capture has been selected. This screen allows for QAP selection. If the QAP name and MAC address are unknown the search function can be used to obtain these details. The options relating to minimum capacities can then be set. From here each AC can be assigned a minimum capacity (*Cmin*) which the controller will endeavour to provide to the AC. Default IEEE 802.11b and IEEE 802.11e EDCA settings can also be set at this point via the radio buttons. Once this screen had been completed the main screen of the program is launched.

The minimum capacity (*Cmin*) input settings entered here are key to the operation of the WRRC. In order to prioritise the video stream it would be very easy to set the *Cmin* allocation of the video AC to 0.90 and share the other 0.10 between the three remaining ACs. This setting would serve to punish other traffic on the network severely and unfairly. In order to employ the most useful RRM strategy for video streaming it is important that the values of *Cmin* are intelligently chosen in order to give a tradeoff between fairness and end user QoE. The experiments outlined in this thesis aim to determine the most effective value of *Cmin* to choose for a range of video contents and characteristics and background load levels.

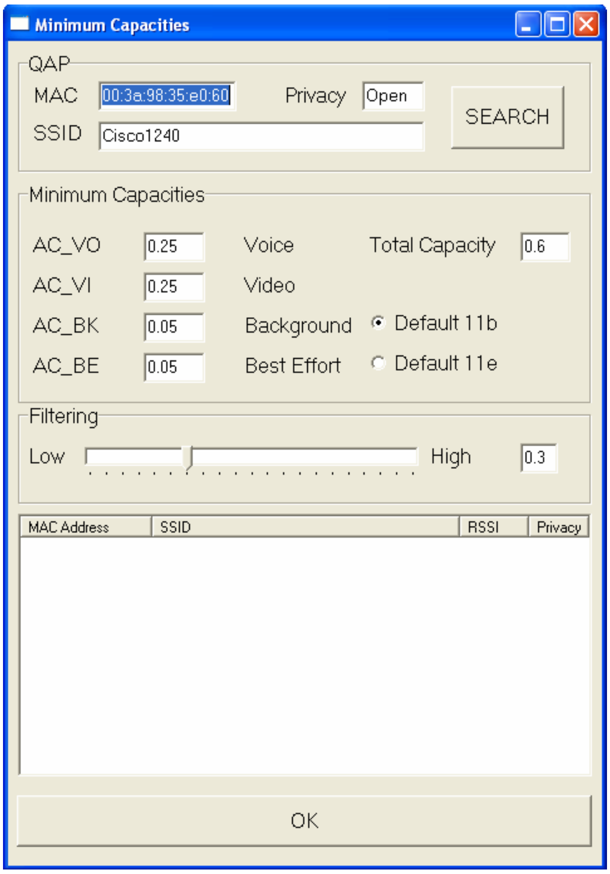

Figure 13: WRRC Minimum Capacities Configuration Window

The CNRI WRRC application has been available for a number of years and a demo version is available from the CNRI website [42]. Based on the *Cmin* values entered by the user, the application modifies the EDCA settings of the IEEE 802.11e protocol to provision bandwidth for each of the ACs. Since it has been developed, work has been undertaken to determine what values of *Cmin* should be used for voice over WLAN (VoWLAN) applications [43]. This application reduces the number of values that must be set by the user to just one for each AC. This thesis aims to determine what values should be used for *Cmin* in order to deliver video quality that meets end user requirements.

The main WRRC window (Figure 14) presents the options to switch between monitor and control mode. There is also an option to record a live stream. A wealth of information is presented on this screen. Channel details such as load, contention and transmission rate are shown. Details relating to individual ACs are also shown in a variety of textual and graphical formats.

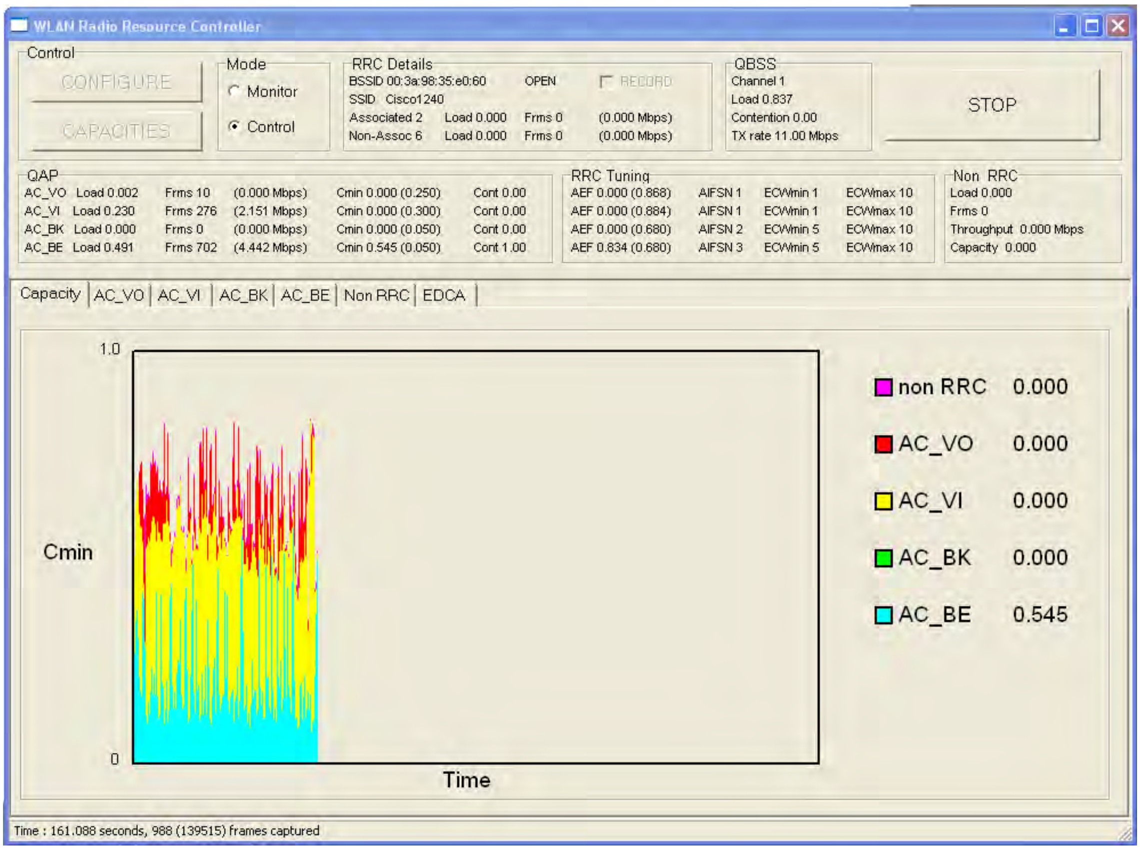

Figure 14: WRRC Main Window

This screen is divided into three rows of panels. The **Control** panel contains two buttons which enable the configuration screens shown in Figure 12 and Figure 13 to be recalled. These buttons are greyed out here as they cannot be recalled while the WRRC is active. The **Mode** panel allows the dynamic EDCA control mechanism of the WRRC to be enabled and disabled, i.e. the control and monitor modes respectively. The **RRC Details** panel gives network information that includes the SSID, MAC address and associated clients. The ability to record a trace of the activity is also available. The **QBSS** panel gives some higher level information on the channel including the current channel number, channel load, the contention level and average transmission rate. The **Stop/Start** button is used to enable or disable the operation of the WRRC application. The **Non RRC** panel gives details of traffic on the channel that is not associated with the current network. This panel provides information on the traffic generated by other users on the channel over which the administrator (and the WRRC) has no control. The **RRC Tuning** panel displays the current EDCA values for each AC and the **QAP** panel provides details of the load on each AC. This includes the overall load for each AC, the number of frames, the current capacity being used and the contention.

In Figure 14, the bottom panel shows how the capacity of the network is currently being used by the four ACs and the non-RRC traffic. It also shows how the capacity has been used in the past number of iterations. It is worth noting that the program can display up to 600 previous measurements.

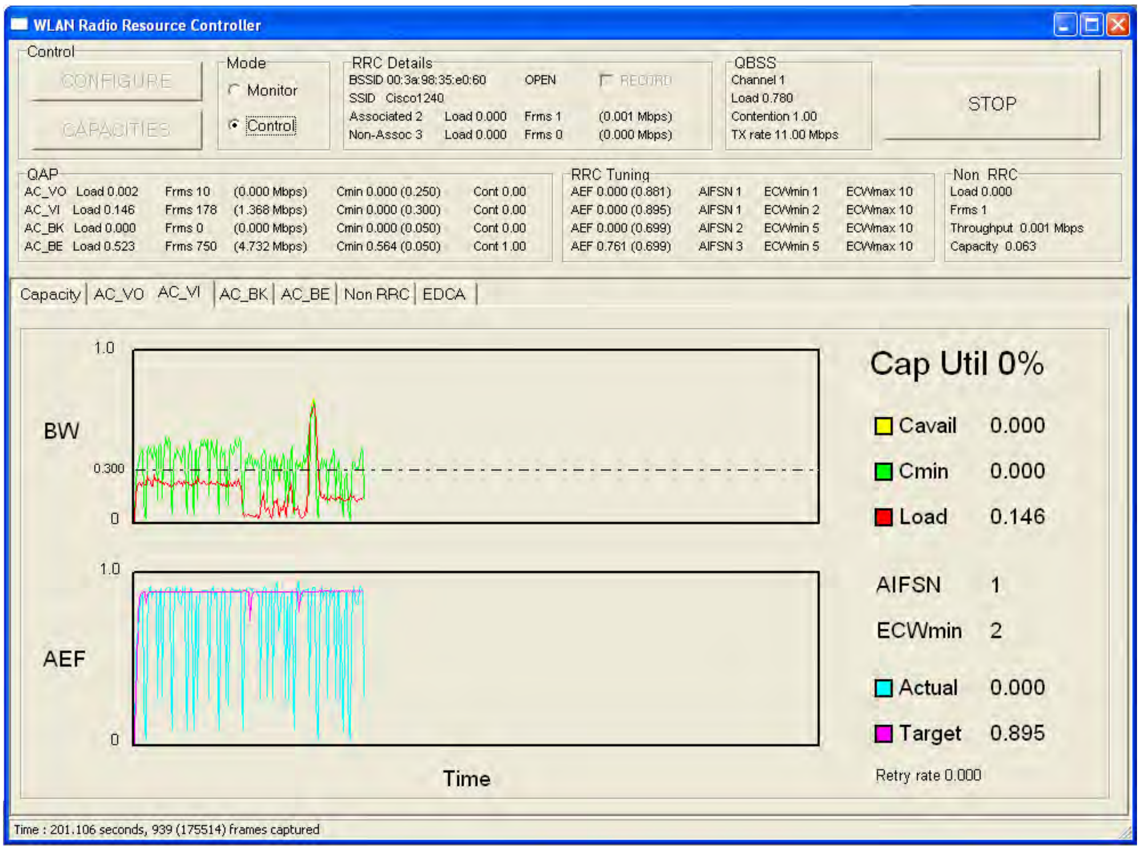

Figure 15: WRRC main window showing Video panel

In Figure 15, the bottom panel of the controller has been switched to display the Video AC performance. This display shows how the load placed on the video AC changes over the running duration. The Green trace on the upper graph shows the capacity (*Cmin*) that was available to that AC and the red trace corresponds to the load placed on that AC. The bottom graph displays the target and actual values for the *AEF*. To the right of the graphs the current state of the AC is displayed in text form. This screen is also available for the Voice, Background and Best Effort ACs and also the Non-RRC traffic by selecting the relevant tab.

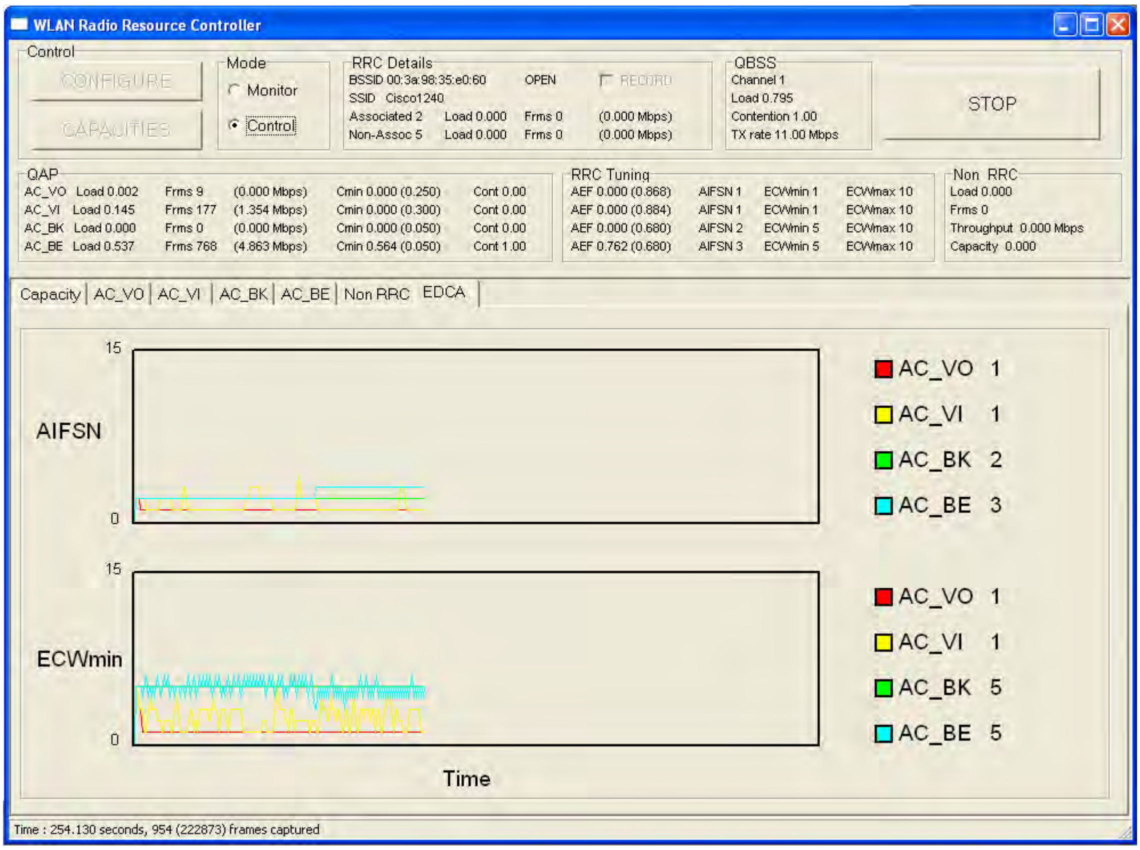

Figure 16: WRRC main window showing EDCA panel

Figure 16 above shows the EDCA tab of the WRRC. This displays how the *AIFSN* and the *ECWmin* values change over time. Each AC is represented by a different colour: red for VO, yellow for VI, green for BK and blue for BE. The current values of *AIFSN* and *CWmin* for each AC are displayed to the right of the graphs.

4.2.2 Video File Preparation

The set of test video clips used in the experiments detailed below were created using the Xilisoft DVD Ripper Pro software. Image sequences from the feature films “Little Miss Sunshine” and “A Scanner Darkly” were chosen as they represent a range of content complexity.

A 10 minute clip from “Little Miss Sunshine” was chosen as it presented a range of different scenes with varying light levels, motion and scene chance frequency. The film “A Scanner Darkly” was chosen as it is an animated movie with strong colours and hard edges. In previous work [44], [45] it has been shown that animated video clips present a larger than normal bandwidth demand on the wireless network.

Both clips were encoded into MPEG-4 format with three separate bit rates: 1500kbps, 2000kbps and 2500kbps. These bit rates were chosen as they are typical for modern video applications. For each bit rate three different resolutions were used; Common Intermediate Format (CIF) 352 × 288 pixels, Quarter CIF (QCIF) 176 × 144 pixels and 4CIF 704 × 576 pixels. All video files had a frame rate of 25fps and had the audio channel disabled. A hint track was added to each of these clips using the mp4creator software. For these experiments a MTU of 1024 bytes was chosen. This hint track is used by the streaming server to determine how the video file should be streamed.

4.3 Objective Video Metrics

4.3.1 PSNR Calculation

Peak Signal to Noise Ratio (PSNR) is a full reference objective metric for determining image quality. It makes use of both the host and client side video files and returns a decibel (dB) value for the video clip. It is calculated on the luminance signal of a video file and is easy to compute.

For 2 video files where *I* is a host side video frame and *K* is a client side video frame, both composed of *i* by *j* pixels, the mean square error is computed according to

$$MSE = \frac{1}{mn}\sum_{i=0}^{m-1}\sum_{j=0}^{n-1}\left[I(i,j) - K(i,j)\right]^2 \qquad (4.1)$$

The PSNR is then calculated according to

$$PSNR = 10\log_{10}\left(\frac{MAX^2{}_I}{MSE}\right) = 20\log_{10}\left(\frac{MAX_I}{\sqrt{MSE}}\right) \qquad (4.2)$$

where MAX_I is the maximum pixel value that can occur., in the case of YUV files this value is 255.

4.3.2 Video Quality Metric (VQM)

The VQM metric [23], [46] is a quadratic mapping scale for PSNR values. It is defined for PSNR values between 20dB and 45dB and converts PSNR values in the range into values between 0 and 1. The VQM of a PSNR value is calculated according to

$$VQM(PSNR) = 0.0007816 \times PSNR^2 - 0.06953 \times PSNR + 1.5789 \qquad (4.3)$$

A VQM of zero implies that there is no impairment between the host and the client side video while a VQM of one indicates maximum impairment.

Unfortunately this metric is only defined for PSNR values between 20 and 45. In order to overcome this restriction a modified VQM metric has been devised for the purpose of this thesis.

The VQM has a value of 1 at a PSNR value of 9.3dB indicating that at this value the maximum impairment has been reached. Therefore all PSNR values at and below 9.3dB correspond to maximum impairment. Similarly, a PSNR value of 45dB yields a VQM value of zero or no impairment, therefore all PSNR values above 45dB are also of maximum quality with no impairment. For this reason and also because PSNR values outside of the 20dB to 45dB range defined in [23] occurred during the experiments the VQM metric was modified to incorporate the lower and higher PSNR values.

The equation employed to calculate the modified VQM is shown below.

$$VQM(x) = \left\{ \begin{array}{l} 1 \quad ,0 \le x \le 9.3 \\ 0.0007816x^2 - 0.06953x + 1.5789 \quad ,9.3 < x \le 45 \\ 0.033 \quad ,45 < x \le 100 \end{array} \right\} \tag{4.4}$$

The modified VQM weighting curve is shown in Figure 17 below. This equation compresses the PSNR values into a smaller range and constrains the high and low quality frames where the differences are not perceivable to the HVS as no impairment and high impairment respectively.

As noted earlier, the minimum PSNR value for acceptable wireless video transmission is 20dB which equates to 0.5 in the VQM scale.

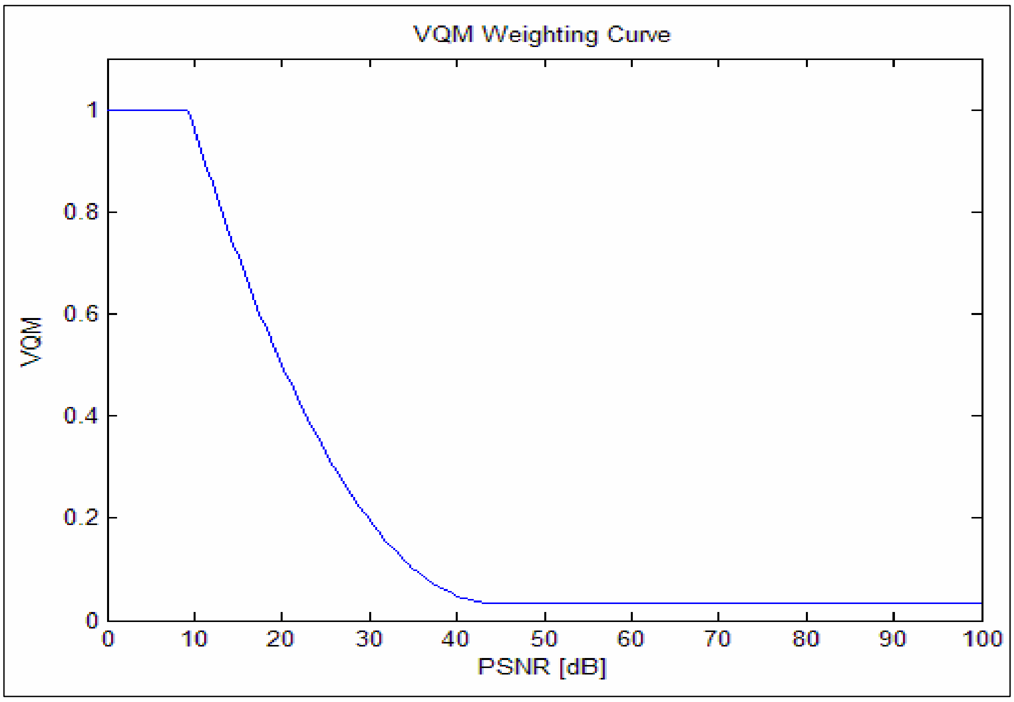

Figure 17: Modified VQM Weighting Curve

The four experimental scenarios investigated in this thesis are described below.

The first experimental scenario (further details in section 4.4) describes how a video transmitted over a cabled Ethernet connection was used to generate a set of benchmark results to evaluate the quality of videos transmitted over the wireless channel.

In section 4.5 the second experimental scenario is described. This is concerned with transmitting video over a wireless connection with various background loads where only a single client is present and receives both the background and video traffic. Tests are conducted using both static IEEE 802.11e and adaptive IEEE 802.11e for comparison.

The third experiment, described in section 4.6, is concerned with video transmission over a wireless connection with a variable background traffic load present. The video traffic and best effort traffic are destined for independent wireless clients.

The fourth experiment is described in section 4.7. For this test the video AC (VI) is allocated a range of minimum capacity (*Cmin*) values by way of the CNRI WRRC (Section 4.2.1). For each capacity level a fixed background traffic load is maintained. The resulting video quality is then compared for each level of capacity allocated. This experiment allows the optimal value of minimum capacity (*Cmin*) to be set for any given video stream.

4.4 Scenario 1 - Analysis of Video Streamed Over Wired Connection

For this test a set of video files with a range of resolutions and bit rates were streamed from the server PC to a wired client via a 24-port switch. The video files were 10 minutes in duration resulting in approximately 15000 video frames being generated.

Video	Bit rate (kbps)	Resolution
Miss Sunshine	1500	QCIF, CIF, 4CIF
Miss Sunshine	2000	QCIF, CIF, 4CIF
Miss Sunshine	2500	QCIF, CIF, 4CIF
A Scanner Darkly	1500	QCIF, CIF, 4CIF
A Scanner Darkly	2000	QCIF, CIF, 4CIF
A Scanner Darkly	2500	QCIF, CIF, 4CIF

Table 9: Video Characteristics for wired connection

The client side videos were played-out and recorded using the VLC media player. The captured video files were then converted from MPEG-4 format to YUV format and the PSNR calculation was performed using the YUVAnalyser software. These values were then further processed using PERL and Excel scripts to generate statistical PDF and CCDF distributions. The video files range in resolution from CIF to 4CIF and in bit rate from 1500kbps to 2500kbps. A large MTU of 1024 bytes was used throughout this test as the video files used have a large bitrate.

As this is a simple client/server topology, it is expected that the resulting client side videos are of a high quality with high PSNR values for each frame and a high average

PSNR for each clip. The resulting PDF of these values should therefore exhibit a large peak at high PSNR values.

The aim of this experiment is to obtain a set of results to act as a benchmark with which to compare the subsequent wireless transmission results. In this case the wired link offers a bandwidth of 1 Gbps and there are no other nodes on the network to affect these results.

4.5 Test Scenario 2 - Experimental Setup

The purpose of this experiment is to compare the quality of a streamed video when it has been transmitted over an IEEE 802.11e network with static EDCA settings compared to when it has been streamed over an IEEE 802.11e network where the EDCA values are dynamically controlled by the CNRI WRRC. For the duration of this experiment the minimum capacity allocated to the Video AC was set to $Cmin = 0.25$.

A video server PC and a traffic generating PC were connected to the Cisco QAP via a 24-port switch. The wireless client PC received both video traffic and best effort traffic via a Netgear PCI wireless adapter (Figure 18). Best effort traffic was placed on the network using the D-ITG traffic generator with a load that varied in both packet size and packet rate according to the table below (Table 10). A Poisson distribution was used to shape the packet rate and packet size over time. This was used to simulate other users on the network consuming bandwidth. The loads placed on the best effort AC were large in order to force the network into a near saturation scenario. This would ensure that the video AC would find it difficult to obtain enough bandwidth for a successful transmission.

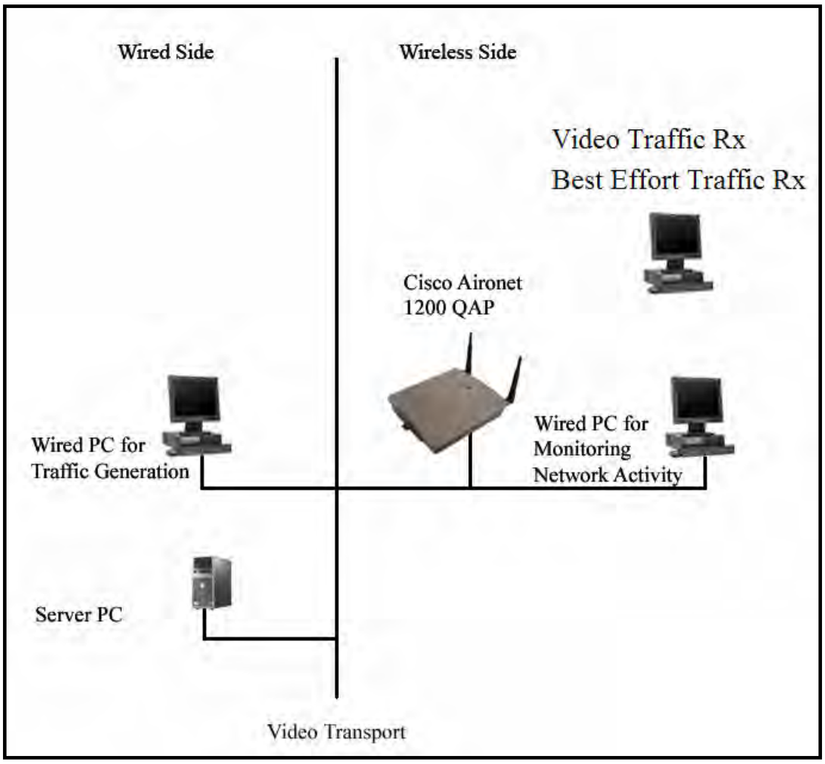

Figure 18: Network topology used for experimental scenario 2

Packet Rate (packets per second)	Packet Size (bytes)
500	128
500	256
500	512
500	768
500	1024
750	128
750	256
750	512
750	768
750	1024

Table 10: Best Effort Traffic Levels

A PC running the CNRI WRRC software and fitted with an AirPcap wireless packet capture device was used to monitor the network when operating in static IEEE 802.11e mode and to monitor the network and dynamically control the EDCA parameters when using adaptive IEEE 802.11e settings. The WRRC was also set to record the network activity for analysis at a later stage.

The VLC media player was used to play out the video on the client side and also to capture the stream to an MPEG-4 file for comparison. The original video files and the captured video files were then converted to YUV format and analysed using the YUVAnalyser software. The excel files generated by YUVAnalyser were then further processed by using a Matlab script to generate the PDF of the PSNR and VQM values and to generate the CCDF of the PSNR values and the CDF of the VQM values. The example below (Figure 19) shows the resulting graphs for a single experimental run. The different traces correspond to different background traffic loads. Clockwise from top left the graphs represent the PDF of the PSNR values, the PDF of the VQM values, the CDF of the VQM values and the CCDF of the PSNR values. The CCDF is chosen for the PSNR values as higher values are more desirable but in the case of the VQM results lower values are more desirable leading to the CDF being utilised.

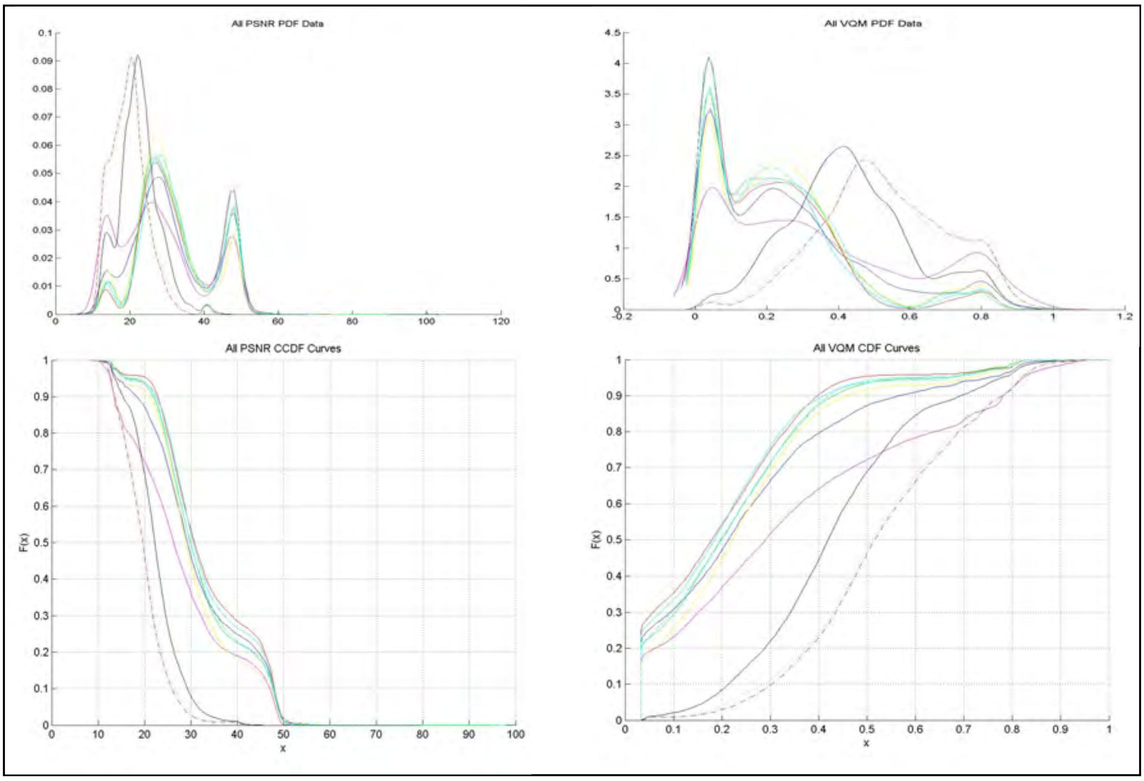

Figure 19: Example of results generated showing PSNR PDF, PSNR CCDF, VQM PDF and VQM CCDF

4.6 Test Scenario 3 - Experimental Setup

This experiment expands on the experiment detailed in section 4.5 and introduces multiple clients into the wireless topology. On the client side of the topology a single client is used to receive video traffic while another wireless client serves as the destination for the best effort traffic. The topology is graphically presented below (Figure 20). The best effort traffic varied in both average packet size and rate according to a Poisson distribution and is injected into the network using the D-ITG traffic generator. The minimum capacity allocated to the Video AC was set at *Cmin* = 0.25 for the purposes of this experiment. The background loads are the same as in the previous experiment and detailed in Table 10. The WRRC was used to monitor and record the network activity when static IEEE 802.11e settings were used and was also used to control the EDCA parameters for adaptive IEEE 802.11e mode. The VLC media player was used to capture and play out the client side video.

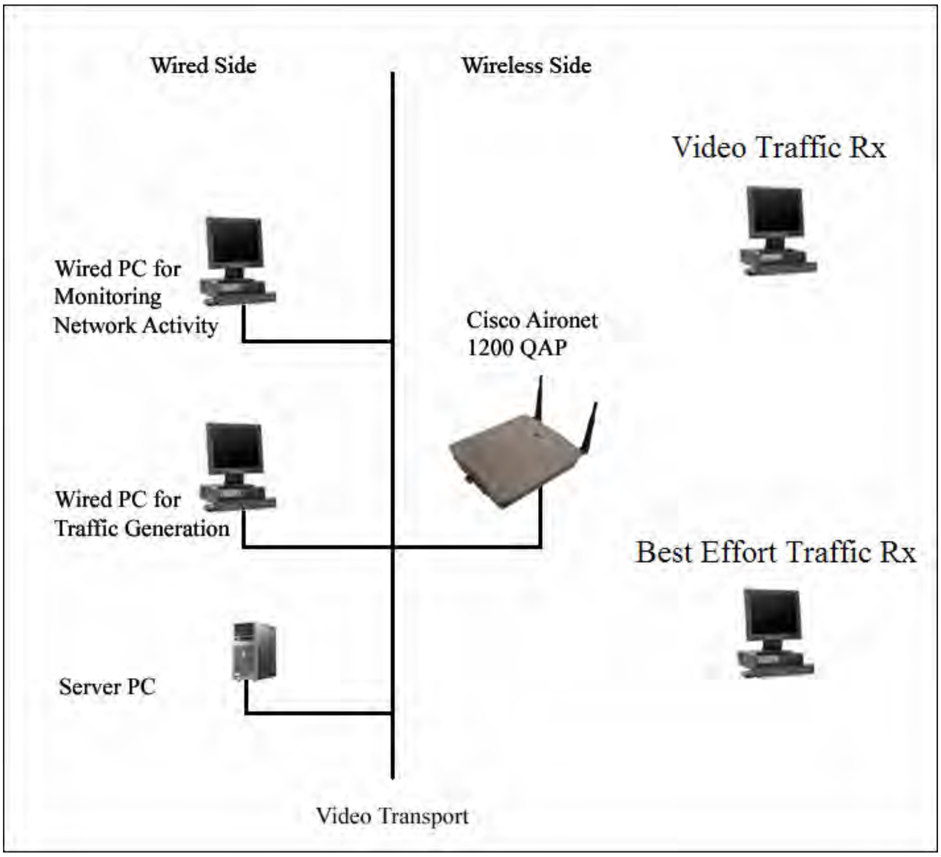

Figure 20: Network topology used for experimental scenario 3

In all cases the host and client side MPEG-4 videos were converted into YUV format utilising ffmpeg and the YUVAnalyser tool was used to generate PSNR values for each clip. These values were processed using Matlab to generate PDF, CDF and CCDF plots.

4.7 Test Scenario 4 – Experimental Setup

This experiment aims to quantify how the quality of a streamed video is affected by the capacity allocated to the video AC and to determine the value of *Cmin* required to deliver acceptable end user QoS. In order to conduct this experiment a subset of video files was selected which contained the video files that showed the greatest and the least improvement in quality in the previous experiments (Table 11). These video files were then transmitted in the same manner as detailed in section 4.6. However in this experiment each video/background load combination was transmitted for different *Cmin* settings where *Cmin* allocated to the video AC was increased from 0.05 to 0.30 in steps of 0.05. The purpose of this test is to determine an optimum value for the allocated video capacity in order to meet end user quality requirements. If the perceived quality does not vary dramatically over a given range of capacities then it is unnecessary to allocate a higher capacity if a lower allocation yields the same quality levels.

Video File	**Bitrate (kbps)**	**Background Load (pps)**	**Background Load (Packet Size, Bytes)**	**Topology**
Miss Sunshine	1500	750	768	1
Miss Sunshine	2000	750	1024	1
Miss Sunshine	2500	750	768	1
A Scanner Darkly	2000	750	1024	1
A Scanner Darkly	2500	750	256	1
A Scanner Darkly	2500	750	768	1
Miss Sunshine	1500	750	1024	2
Miss Sunshine	2000	500	1024	2
Miss Sunshine	2000	750	768	2
Miss Sunshine	2000	750	1024	2
Miss Sunshine	2500	750	1024	2
A Scanner Darkly	2000	750	1024	2
A Scanner Darkly	2500	750	256	2
A Scanner Darkly	2500	750	768	2

Table 11: Video files, Background Loads and Topologies used

5.1 Experiment 1 – Video Streaming Over a Wired Network

During the course of this test it became apparent that the results obtained were not those expected. The YUVAnalyser program stores the PSNR values for each frame of the video clip in an Excel document. When it has completed an evaluation it presents the user with the average PSNR for the entire clip. It was expected that this average would be close to the maximum achievable PSNR value of 100 dB. If the client side and host side videos are visually identical then a PDF of the PSNR values obtained by comparing the videos should exhibit a large peak about the maximum PSNR value as shown below in Figure 21.

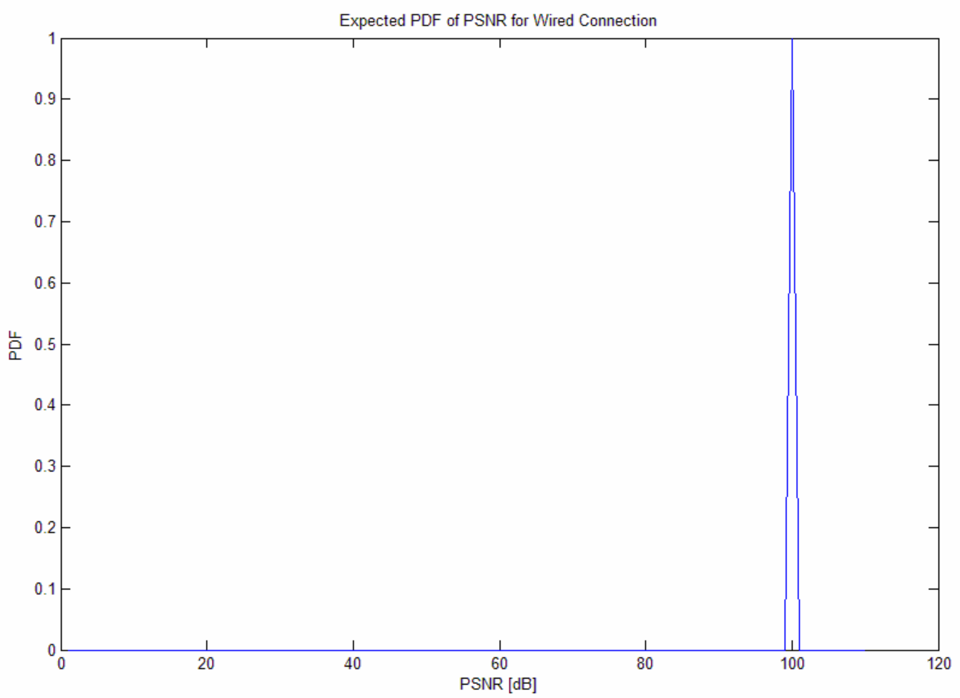

Figure 21: Expected PDF of PSNR values

It was observed after the initial experiments that the average PSNR values were generally in the range of 20dB to 30dB. As the average is only a first order analysis it was apparent that a more detailed statistical analysis was required. A PDF of the PSNR values (Figure 22) showed that initial results were not consistently high as had been expected but in actuality ranged from approximately 8dB up to approximately 46dB with a main peak between 20dB and 30dB and a secondary smaller peak at approximately 14dB.

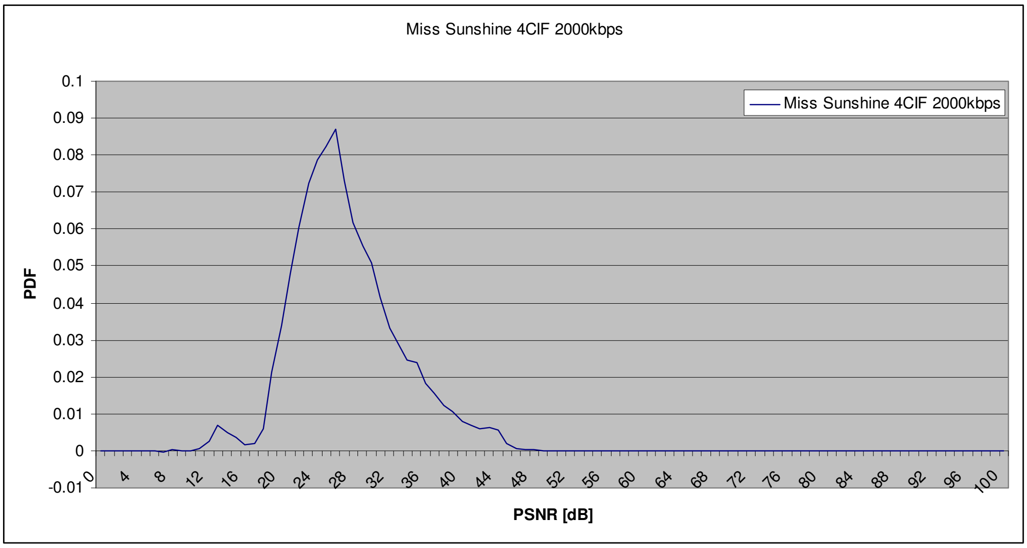

Figure 22: PDF of PSNR values for video streamed on a wired connection

Given that this experiment was carried out on a wired test bed (using Ethernet connections) with no external interference, it was expected that the majority of frames would have a PSNR value approaching 100dB. However as can be seen from the figure above, the largest value of PSNR achieved for this test was approximately 50dB.

There are three possible explanations as to how this distribution in PSNR values could have arisen: network losses, video encoder and video server. Packet loss on the medium could lead to a result similar to this but given that this experiment was carried out on a wired 1Gbps network with low packet loss levels this would have to be ruled out. The encoding standard used at the client side device was set to MPEG-4 using MPEG-4 encapsulation which is the same as that used on the host side server ruling this possible explanation out.

This leaves Darwin Streaming Server as the most likely source of the distribution. There is possibly a bandwidth reduction mechanism in DSS that drops some data from the stream or employs a “code puncturing” algorithm resulting in the sub-optimal video transmission. Darwin Streaming Server was initially available as an open source version of the Quicktime Streaming Server, however, documentation is currently poor.

Figure 23 shows the distribution of PSNR values when both the bit rate and resolution are varied. In this particular example the Little Miss Sunshine video clip was used. It is apparent in this figure that the peak PSNR observed was to be found between 20dB and 30dB, with a smaller secondary peak at approximately 14dB occurring for all cases considered.

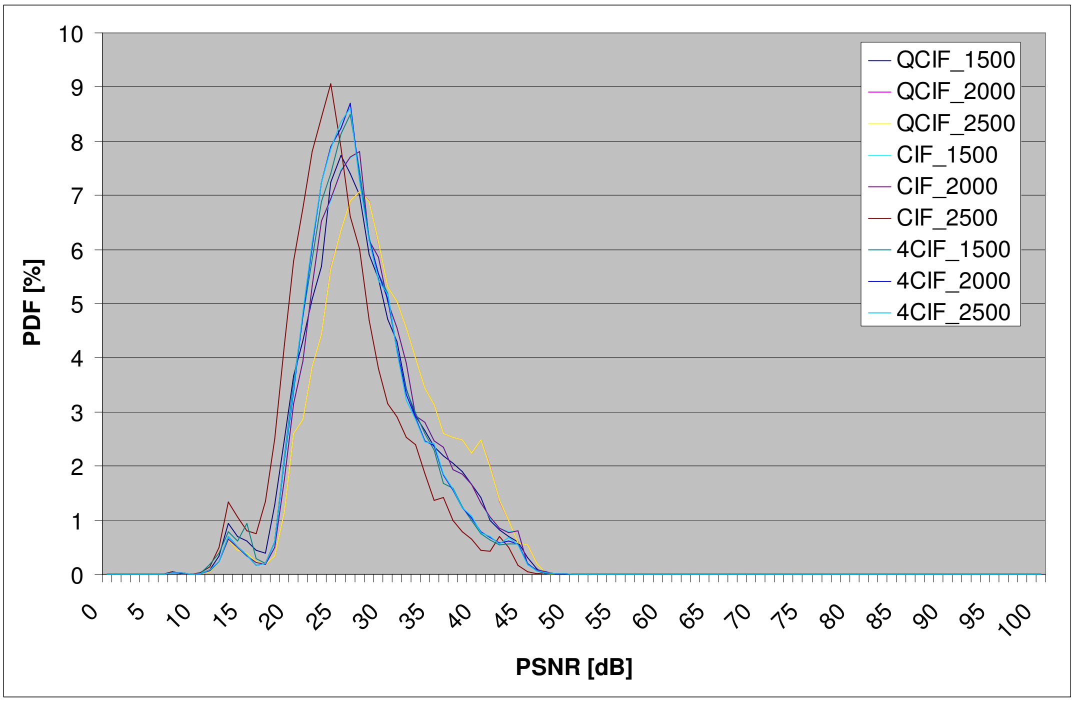

Figure 23: PSNR values Little Miss Sunshine varying resolution and bitrate

The key conclusion from this result it that there is a distribution of PSNR values for video files streamed using DSS from approximately 7dB up to 50dB with two main features occurring, namely a large peak between 20dB and 30dB and a secondary smaller peak at approximately 14dB. It is believed that this distribution pattern is caused by the Darwin Streaming Server application. Frames that have a PSNR value of less than 20dB are deemed unacceptable for wireless transmission and are therefore of little significance since the video is unwatchable. As the PSNR value of a frame increases past 45dB the differences become less detectable to the HVS. For this reason the modified VQM curve was developed and implemented. The main implication that this result presents is that the quality of the video stream has a distribution of PSNR values over a large range even when transmitted under ideal conditions. For this reason a statistical analysis of the results is necessitated as the mean or average results do not give enough information about the video quality.

5.2 Experiment 2 - Comparison of Static IEEE 802.11e Versus Dynamic IEEE 802.11e Networks for Video Streaming With A Single Client

The main objective of this experiment was to quantify the effect that dynamic EDCA control had on video quality when a wireless topology was utilised. For each video file/background traffic level the experiment was carried out twice, once with the static IEEE 802.11e values for EDCA and once with dynamic EDCA control where the video AC was allocated a *Cmin* of 0.25. This value was chosen as it was observed in previous experiments that this level of bandwidth allocation should accommodate the load placed on the network by the video stream.

Figure 24 below shows the CDF of the VQM values obtained when video file *Little Miss Sunshine* with a resolution of 720x576 pixels and a bit rate of 2000kbps was transmitted over an IEEE 802.11e network with static EDCA values where a background load of 750 packets per second and a packet size of 512 bytes was present. The threshold for acceptable quality of 20dB in PSNR equates to a VQM of 0.5, this threshold has been highlighted on the graphs. As can be seen in this experiment using static EDCA values, approximately 46% of the video frames are below this threshold.

Also shown is the result when the CNRI WRRC has been utilised to dynamically control the EDCA settings. The same video bit rate and resolution and the background load are maintained and *Cmin* = 0.25. Here it can be seen that the introduction of dynamic control has increased the proportion of video frames that fall below the VQM = 0.5 threshold to approximately 92%, roughly doubling the amount when compared to static EDCA settings.

More formally,

$$P_{static}[VQM < 0.5] = 0.46$$

$$P_{dynamic}[VQM < 0.5] = 0.92$$

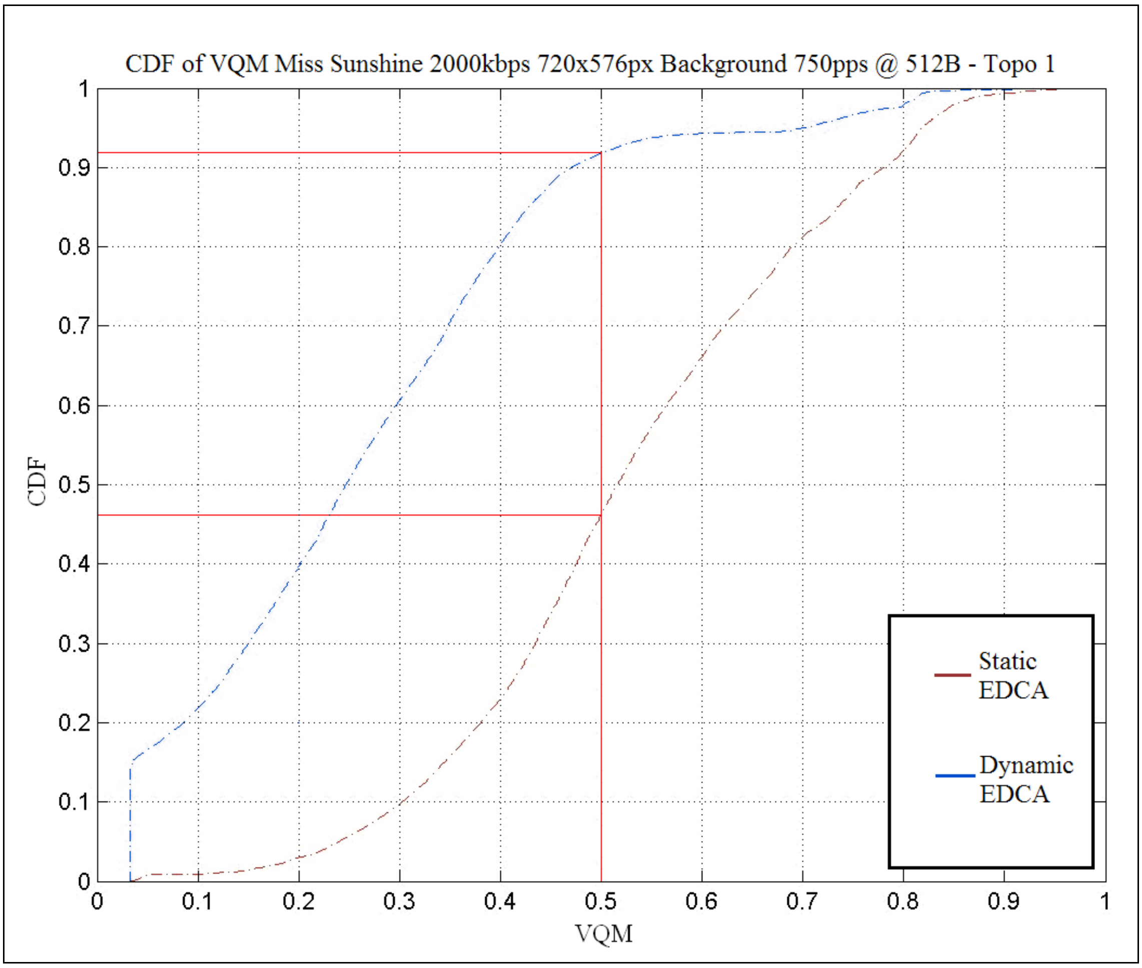

Figure 24: CDF of VQM Miss Sunshine Topology 1 Dynamic EDCA vs. Static EDCA

The video file *A Scanner Darkly* has been chosen as it is an animation video. It has been previously reported that animated clips place the largest bandwidth demand on the network due to their high spacio-temporal dependencies and volume of hard edges and lines. It is therefore expected that lower quality results will be obtained when compared to the *Little Miss Sunshine* video file which is composed of natural images. In this regard *A Scanner Darkly* represented the most challenging scenario for these experiments.

Figure 25 below displays the comparison between static EDCA and dynamic EDCA control for the video file *A Scanner Darkly*. In both tests the bit rate and resolution remain constant at 2000kbps and 720 x 576 pixels respectively. The background load is also maintained at 750 packets per second and 512 bytes per packet. These values are equivalent to the previous experiment where the video file *Miss Sunshine* was transmitted.

It is immediately clear from Figure 25 that the proportion of frames that meet the quality threshold has decreased. With static EDCA values only 20% of the video frames meet this threshold. This is compared with 46% in the case of *Little Miss Sunshine*. As the only variable to change is the video file itself, this degradation can be accounted for by the increased demand placed on the network associated with transmitting an animated video file.

The effect of dynamically tuning the EDCA values for this experimental scenario is also shown. In this case it can be seen that by employing dynamic EDCA the proportion of video frames that meet the quality threshold is increased to 50%. In the case of *Little Miss Sunshine* the static EDCA result was doubled by introducing dynamic EDCA control, in this case the initial result has been increased by more than double and is closer to an increase in the order of 2.5 times more video frames meeting the quality requirement.

Little Miss Sunshine

$$\mathrm{P}_{static}\left[VQM < 0.5\right] = 0.46$$

$$\mathrm{P}_{dynamic}\left[VQM < 0.5\right] = 0.92$$

A Scanner Darkly

$$\mathrm{P}_{static}\left[VQM < 0.5\right] = 0.20$$

$$\mathrm{P}_{dynamic}\left[VQM < 0.5\right] = 0.50$$

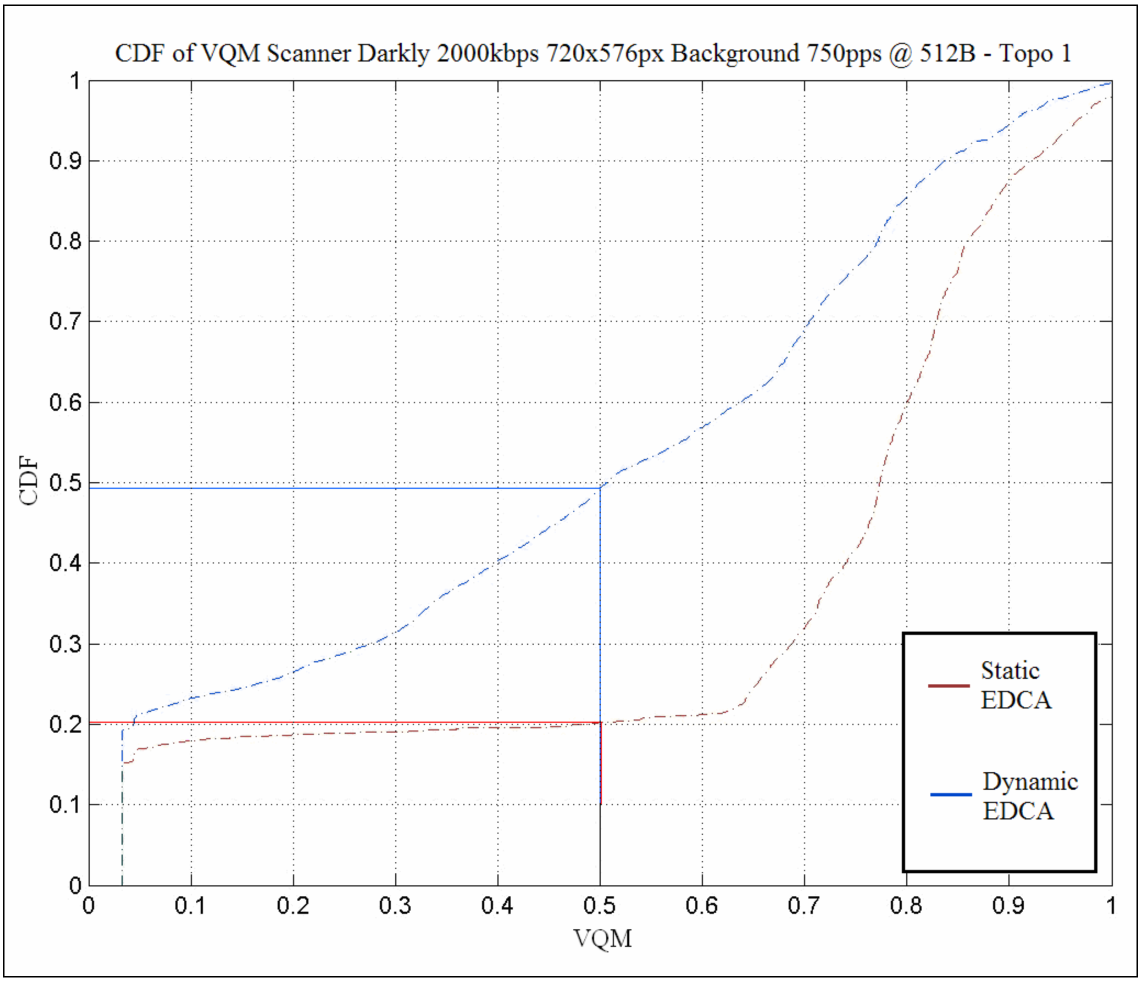

Figure 25: CDF of VQM Scanner Darkly Topology 1 Dynamic EDCA vs. Static EDCA

5.3 Experiment 3 - Comparison of Static IEEE 802.11e and Dynamic IEEE 802.11e Networks for Video Streaming With Multiple Clients

In this experiment a second client was introduced into the topology to act as the destination for the background traffic. By doing this the video traffic was destined for a single client while the background traffic was destined for a second dedicated client. This scenario represents a situation where multiple users are present on a wireless network where one is streaming a video file and the other is downloading a file.

The CDF of the VQM values where static EDCA values were used is shown below (Figure 26). It can be seen here that approximately 62% of the video frames fall within the acceptable threshold for wireless transmission.

Comparing this result to the results presented in the previous section (Section 5.2) there is an increase in video quality when a second client has been introduced to act as the destination for the background traffic. In this case there is a 15% increase in the amount of frames that fall below the VQM = 0.5 threshold when compared to the previous topology.

The effects of employing dynamic EDCA tuning topology 2 can be seen in figure 26 below. In this case the proportion of frames that fall below the VQM = 0.5 threshold has been increased to approximately 95%. This represents a 33% improvement over the static EDCA settings despite the fact that the background load has been maintained at 750pps with a packet size of 512 bytes.

$$\mathrm{P}_{static}[VQM < 0.5] = 0.62$$

$$\mathrm{P}_{dynamic}[VQM < 0.5] = 0.95$$

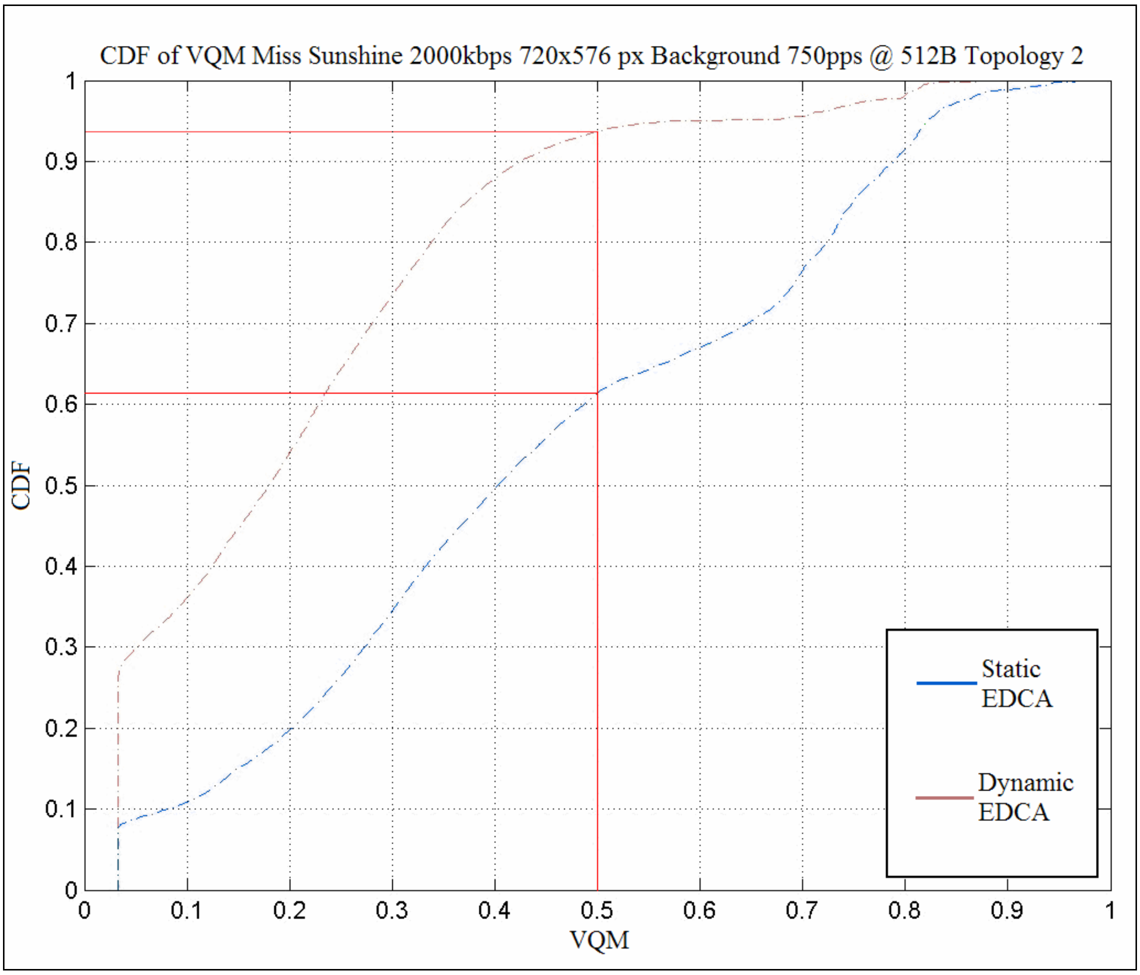

Figure 26: CDF of VQM Miss Sunshine Topology 2 Dynamic EDCA vs. Static EDCA

In the case of *Scanner Darkly*, the introduction of a second client has little effect on the proportion of frames meeting the quality requirement when static EDCA settings are used. It can be seen in Figure 27, that at a VQM of 0.5, 21% of the frames transmitted meet the quality requirement. In this case the background load has been maintained at 750 pps with a packet size of 512 bytes from experiment 2 (section 5.2). This would suggest that with this particular load, video file and these static EDCA settings, this quality level is the best that can be achieved.

The introduction of dynamic EDCA control does lead to an increase in the proportion of frames that meet the quality requirement as can be seen in figure 27. In this instance dynamic EDCA control leads to 41% of the video frames falling below the VQM threshold. Comparing this result to topology 1 where approximately 50% of the frames met the quality threshold with dynamic EDCA control there is a decrease in the proportion of frames that occur within the quality threshold.

In the case of *A Scanner Darkly*:

$$\mathrm{P}_{static}[VQM < 0.5] = 0.21$$

$$\mathrm{P}_{dynamic}[VQM < 0.5] = 0.41$$

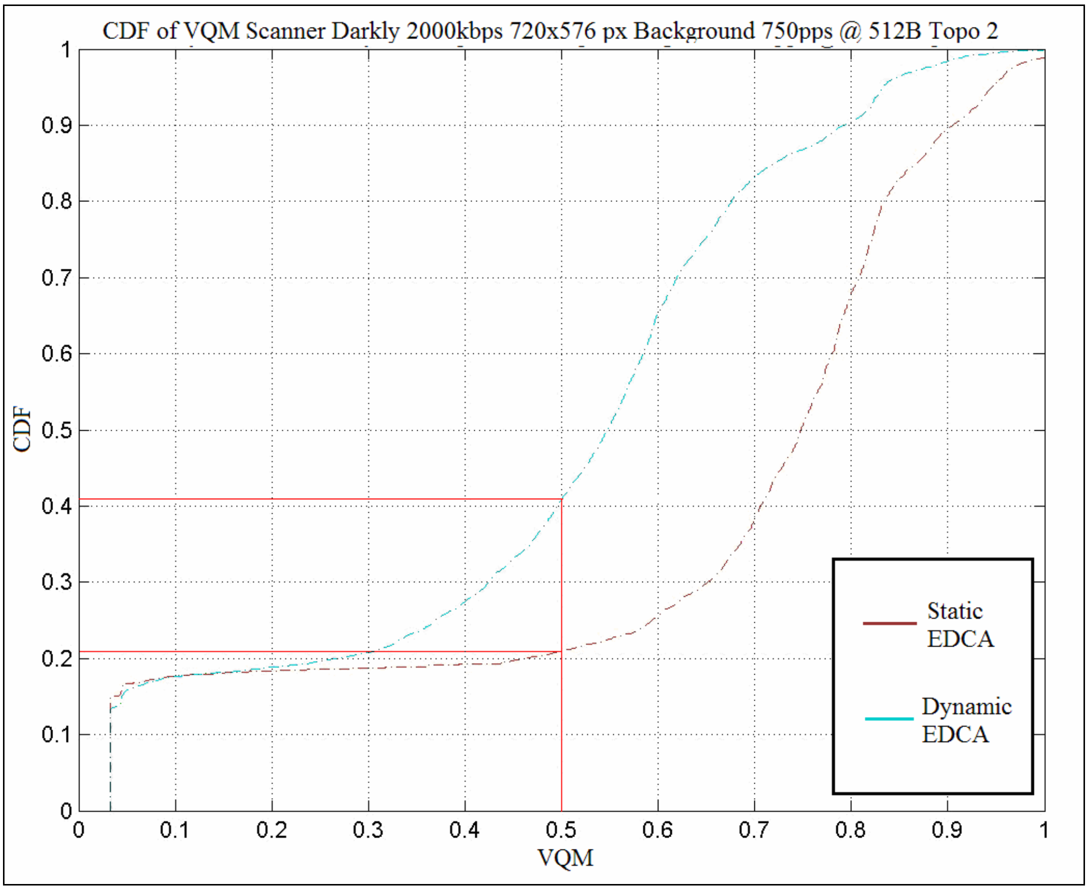

Figure 27: CDF of VQM Scanner Darkly topology 2 dynamic EDCA vs. static EDCA

5.4 Experiment 4 - Effect of Capacity Allocation On Video Stream Quality Over Dynamically Controlled IEEE 802.11e Networks

The purpose of this experiment is to determine the optimum capacity to allocate to the video access category in the WRRC application in order to meet user QoE requirements. It can be seen in Figure 28 that the video streams corresponding to an allocated *Cmin* of 0.05, 0.10 and 0.15 all fall below 40% of frames meeting the quality requirement. The video streams where *Cmin* was set to 0.20, 0.25 and 0.30 where all have above 85% of their frames meeting the quality threshold. In these instances the background load is set to 750 pps with a packet size of 768 bytes.

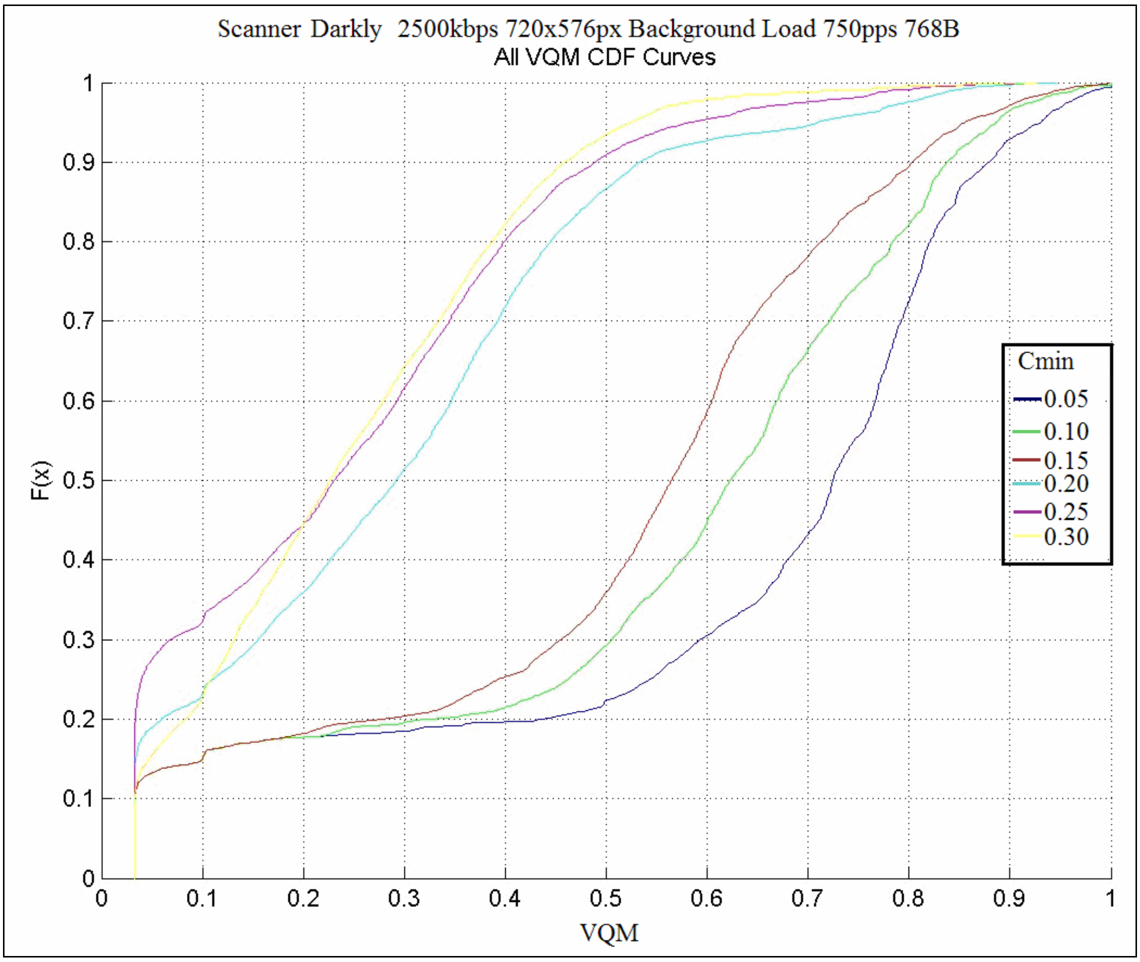

Figure 28: Effect of capacity allocation: Scanner Darkly 2500kbps Background 750pps 768B

Figure 29 displays the results when the packet size has been changed to 256 bytes. All other variables have been maintained. In this case it can be seen that allocating a *Cmin* of 0.20 to the video AC results in approximately 30% of the video frames meeting the quality requirement. The reduction in packet size reduces the *AEF* for the best effort AC as seen in section 4.2.2. As the best effort AC needs to access the medium more frequently the video AC has suffered. This has led to a 55% reduction in acceptable quality video frames.

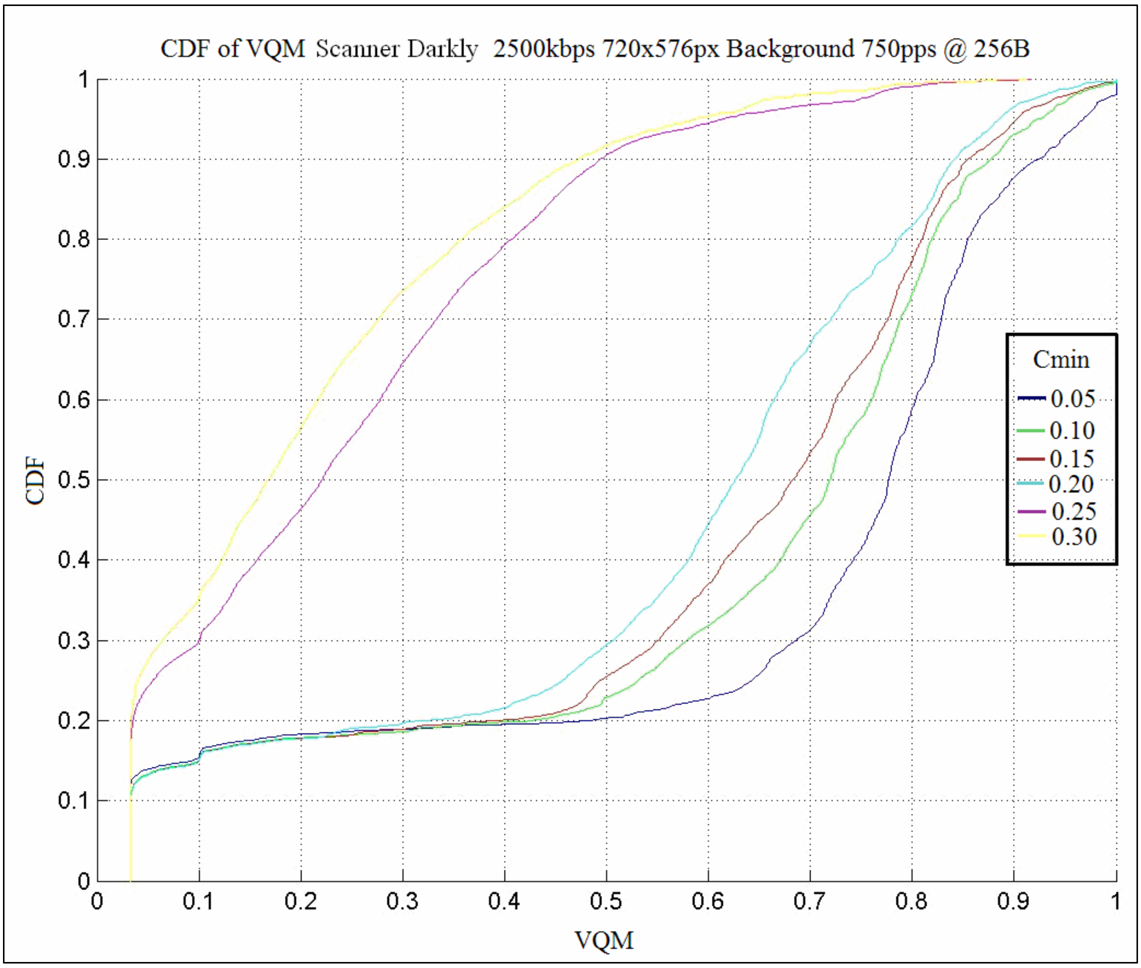

Figure 29: Effect of Capacity allocation: Scanner Darkly 2500kbps Background 750pps 256B

In Figure 30, the results are displayed when the video file *Little Miss Sunshine* with a bit rate of 2000 kbps and a resolution of 720 x 576 pixels was transmitted where a background load of 750 pps with a packet size of 1024 bytes was present. In this case all traffic is destined for a single client node. It can be seen that there is a significant increase in the proportion of video frames that satisfy the quality threshold from 28% of the frames with a *Cmin* = 0.05 to approximately 93% of the frames when a *Cmin* = 0.10 is allocated. Once this proportion of frames has been achieved there is no significant increase in the quality when additional capacity is allocated to the AC. For this scenario there is no benefit to allocating any more than *Cmin* = 0.15 to the video channel. It can also be seen that with a *Cmin* allocation of 0.30 there is a slight drop in the proportion of frames that meet the quality requirement.

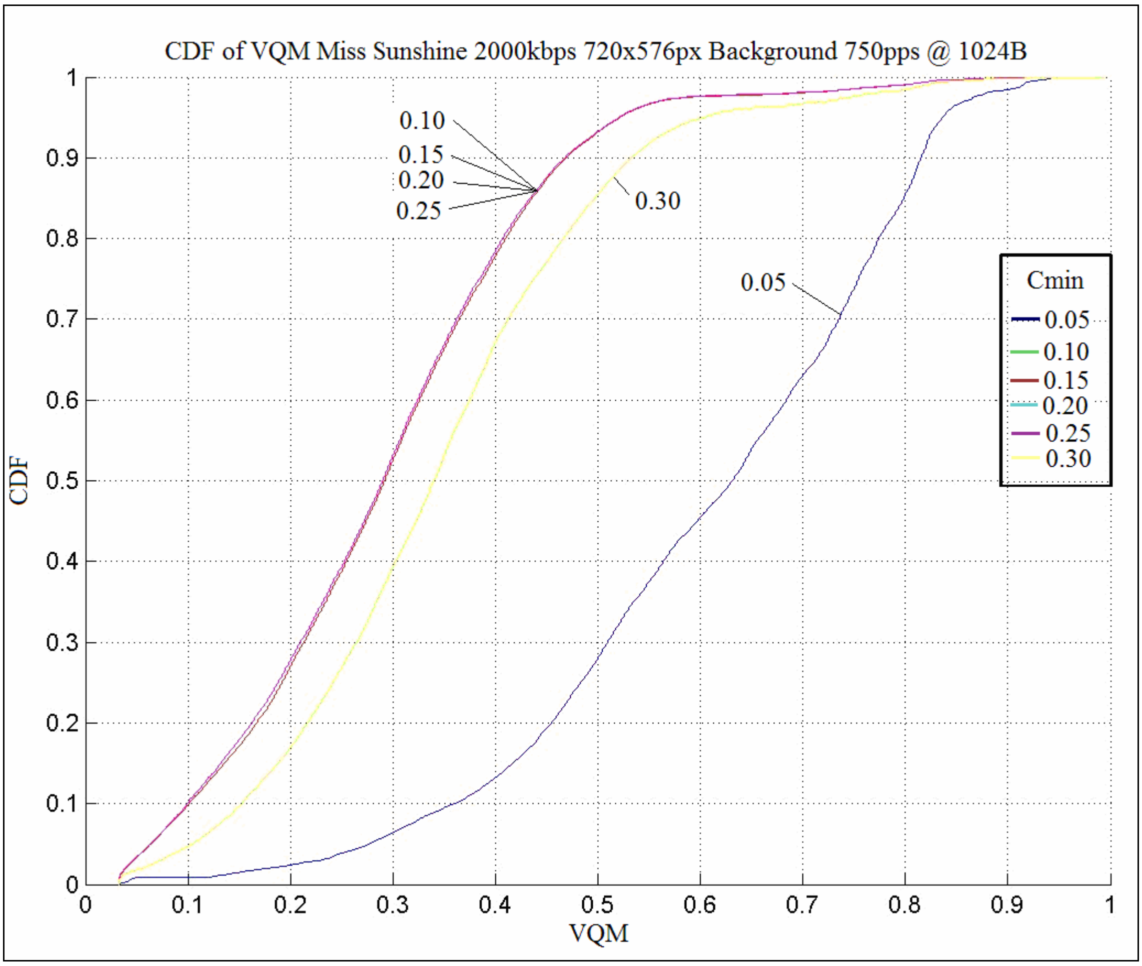

Figure 30:CDF of VQM Miss Sunshine 2000kbps Background 750pps 1024B Topology 1

Comparing this result to that obtained when the second topology was used (Figure 31) and a *Cmin* allocation of 0.05 was set, the video quality has improved dramatically. In the case of topology 1, approximately 28% of the video frames were within the quality threshold and provided the poorest quality results out of all the capacities tested. In the case of topology 2, the proportion of video frames falling within the quality threshold with a *Cmin* of 0.05 allocated to the video AC has improved to approximately 98%. This is the highest proportion of frames to meet the quality requirement for all the capacities tested. Allocation of additional *Cmin* has resulted in less frames meeting the quality threshold and with the highest *Cmin* allocation of 0.30 the lowest proportion of frames meet the quality requirement.

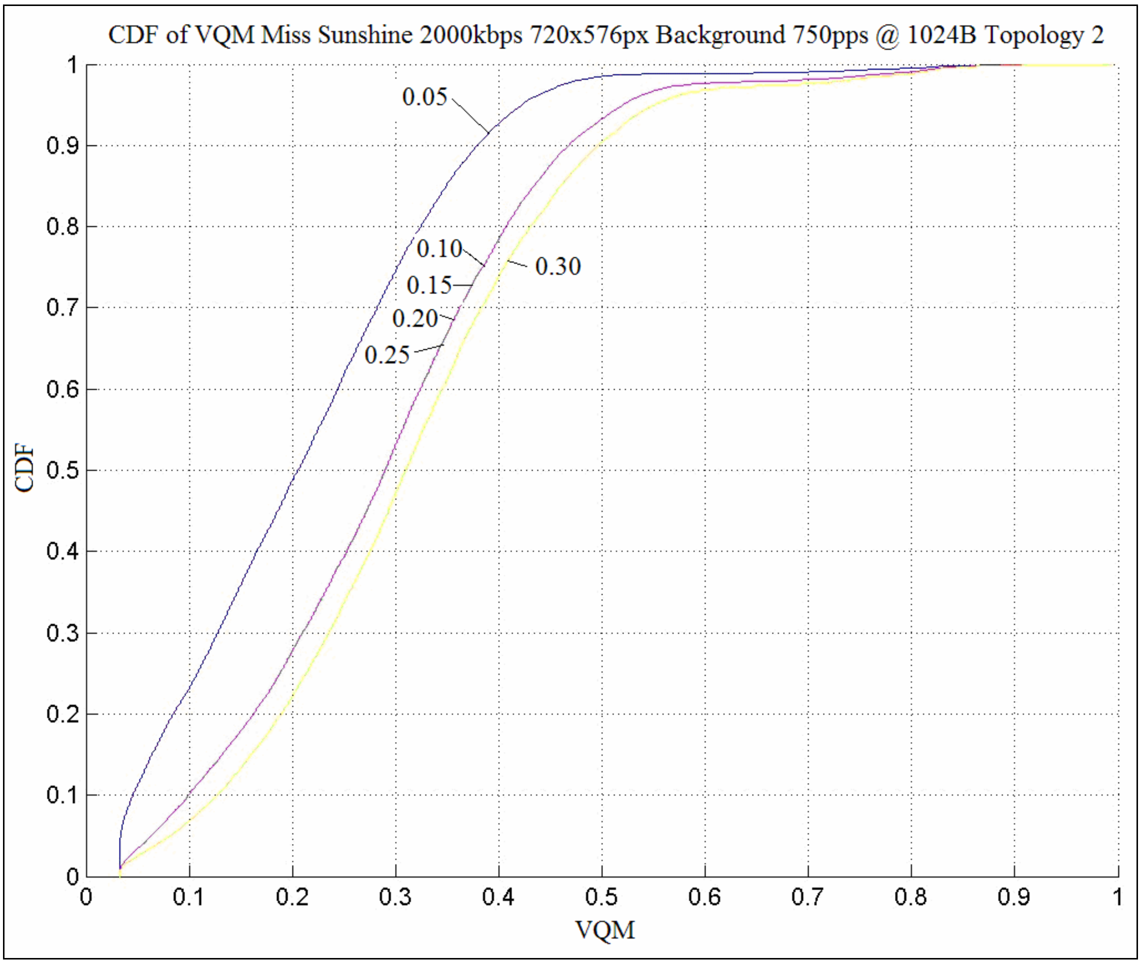

Figure 31: CDF of VQM Miss Sunshine 2000kbps Background 750pps 1024B Topology 2

Further investigation is required into why the quality drops off towards the higher end of capacity allocation but an initial investigation points towards a minor oversight in the control algorithm employed by the WRRC application.

When setting up the WRRC (Figure 13), the user is prompted to enter four capacity values corresponding to the minimum capacity to be allocated to each of the ACs, however, as the medium is shared there are instances when not all of the capacity will be available. This can lead to the WRRC attempting to realise AEF values that are unattainable in real networks under these circumstances. This in turn causes inappropriate EDCA values to be sent to the AP leading to reduced quality.

Figure 32, below, illustrates how the video quality changes with capacity allocation for this particular scenario. It is clearly seen here that for topology 2, higher quality is attained for lower capacity allocations and also that the quality doesn't suffer as drastically with the highest capacity allocation.

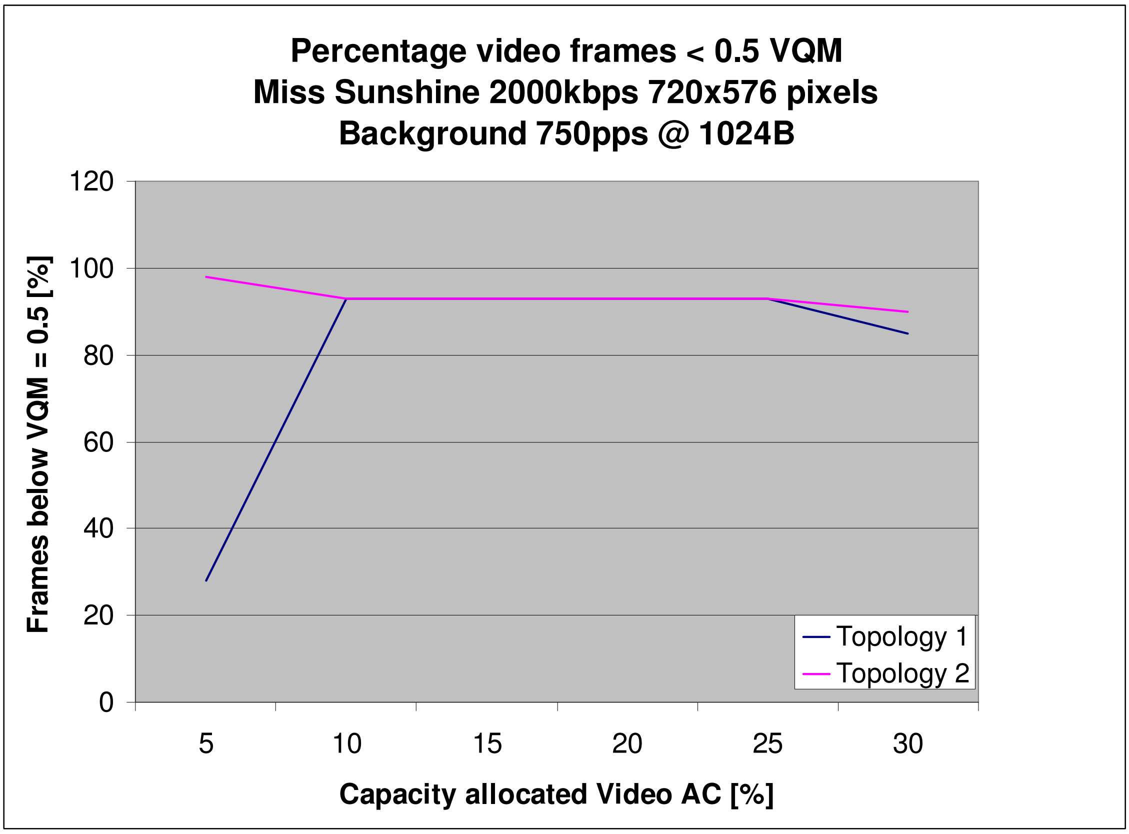

Figure 32: Video Quality vs. Capacity Miss Sunshine 2000kbps Background 70pps 1024B

5.5 Summary

Streaming a video file over a wired network in order to generate a set of best-case results illustrated that the PSNR of the frames arriving at the client were not close to the highest attainable values. As the host and client side video files should be an exact match, the PSNR values should be maximal. Instead it was observed that the PSNR of the received frames had a distribution around 28dB. This discovery led to implementing a statistical analysis of the subsequent results. The cause of the distribution was thoroughly researched with the conclusion that the Darwin Streaming Server elects not to transmit some data relating to the video file. This test also demonstrated that a minimum capacity of 0.25 would be a good initial value for following experiments.

In the case where dynamic RRM was employed over an IEEE 802.11e network a clear performance increase in terms of QoE was observed when compared to static EDCA values. It was also demonstrated that the additional bandwidth required by animated video files led to a lower proportion of frames meeting the quality requirement when compared to natural imagery. Typically one third less improvement was observed when animation was transmitted. The percentage improvements observed are displayed below.

Topology	**Video *Cmin***	**Video File**	**Bitrate [kbps]**	**Resolution [pixel x pixel]**	**Background [pps]**	**[bytes]**	**Improvement P[VQM < 0.5]**
1	0.25	Miss Sunshine	2000	720 x 576	750	512	46%
1	0.25	Scanner Darkly	2000	720 x 576	750	512	30%
2	0.25	Miss Sunshine	2000	720 x 576	750	512	33%
2	0.25	Scanner Darkly	2000	720 x 576	750	512	20%

Table 12: Improvement in P[VQM < 0.5]

From the test results obtained with a *Cmin* value of 0.25 a subset of scenarios was selected to examine when different *Cmin* values were used.

These tests demonstrated that the packet size used for the background traffic has an effect on the proportion of video frames that meet quality requirements. Larger packets result in more video frames falling within the threshold (Table 13). It was also demonstrated that different topologies lead to different proportions of frames meeting the quality requirement (Table 14).

It was also shown that with this RRM application there wasn't a continual increase in the proportion of frames that met the quality requirement as the *Cmin* was increased. Towards the higher levels of *Cmin* allocation a decrease in the number of frames that met the quality requirement was observed. This was attributed to a shortcoming in the control algorithm which is currently being remedied by the CNRI.

Scanner Darkly 2500 kbps 720 x 576 pixels Background 750 pps		
	P[VQM < 0.5]	
Cmin	**768 bytes**	**256 bytes**
0.05	22%	20%
0.10	30%	23%
0.15	36%	26%
0.20	86%	30%
0.25	91%	90%
0.30	94%	91%

Table 13: Variation in P[VQM < 0.5] with packet size.

Miss Sunshine 2000 kbps 720 x 576 pixels Background 750 pps 1024 bytes		
	P[VQM < 0.5]	
Cmin	**Topology 1**	**Topology 2**
0.05	28%	98%
0.10	94%	94%
0.15	94%	94%
0.20	94%	94%
0.25	94%	94%
0.30	85%	90%

Table 14: Variation in P[VQM < 0.5] with topology.

CHAPTER 6 SUMMARY and CONCLUSIONS

6.1 Summary

This thesis has presented an experimental investigation of provisioning bandwidth on an IEEE 802.11e WLAN to improve end-user QoE by employing radio resource management to dynamically tune the EDCA settings based on the current network conditions and user specified minimum capacity allocations. In order to implement RRM, the CNRI WRRC application was employed.

The hostile nature of the wireless medium with limited bandwidth, high loss rates, fading channels and susceptibility to interference from external sources does not suit the non-uniform over time, or bursty, bandwidth profile of a video stream. The real time nature of a video stream also makes error correction by means of data retransmissions more difficult.

The IEEE 802.11e standard provides for four separate transmission queues (or ACs) that can be allotted different levels of access opportunities to the wireless medium. Although this approach cannot provide any further bandwidth it does provide a means to control the allocation of the available bandwidth to the different queues. This method can go some way to overcoming the bandwidth demand of video streaming on the wireless network although it has been generally seen that the EDCA settings that are related to access opportunity have not been used optimally. Generally the EDCA parameters have been statically set to values selected without any clear explanation provided. In order to take full advantage of the protocol it is imperative that the EDCA settings are dynamically tuned along with the dynamic characteristics of the wireless network.

Implementing RRM on a real wireless testbed where real video streaming is taking place is not a trivial task. For that reason the main body of work in the area focuses on computer simulation results where the NS-2 simulator has been commonly used, although the OPNET simulator has been used for a small proportion of these works.

For this thesis, experiments have been designed and implemented on a wireless testbed where real video has been streamed over an IEEE 802.11e enabled WLAN. The video has then been recorded at the client side of the topology and analysed by way of comparison to the original host side video file.

In order to simulate other clients on the network, background traffic was introduced to the wireless network using the D-ITG traffic generator. A range of traffic patterns were chosen that varied in packet size and frequency. A Poisson distribution was used to shape both packet size and frequency to model real network traffic.

Video files were encoded using the MPEG-4 standard and a range of bitrates and resolutions were chosen. The video content encoded presented a range of applications with natural imagery and animation. As shown in previous work, animated video presents a larger demand on a wireless network in terms of bandwidth as it contains large amounts of line art and hard edges.

Any corruption of the stream during transmission can manifest itself at the client side by producing jerkiness, screen freeze and block artefacting, all reducing the users quality of experience. As an end-user’s quality of experience is directly related to their perception of the video, the most accurate quality scores are obtained by carrying out live human trials where participants rate the video they see. This method is both time consuming and costly and must be carried out in a strictly controlled environment. These disadvantages led to the development of objective metrics that aim to estimate the scores generated through live trials.

For the purposes of the experiments in this thesis, the PSNR and a modified version of the VQM have been employed to quantify the end user QoE. The VQM metric has been modified in order to take account of the entire range of PSNR scores available and to reflect the characteristics of the human visual system more accurately.

Through these experiments it has been shown that correct provisioning of the wireless bandwidth can lead to a 2.5 fold improvement in the proportion of video frames that meet end user QoE requirements when compared to static EDCA settings. It has also been shown that with this RRM application that there is not a simple relationship between the bandwidth allocation and the proportion of frames that meet QoE requirements. In fact, over allocation of the bandwidth can lead to a drop in the proportion of video frames that meet QoE requirements. For this reason it is important that the optimal bandwidth allocation is used in order to free up capacity for other applications. This shortcoming of the WRRC algorithm was identified during these experiments and work is underway to correct the algorithm and eliminate the negative effect of capacity over provisioning.

The increase in availability of WLAN access networks is occuring in conjunction with a dramatic increase in the demand for video. According to a Cisco market report 57% of all consumer Internet traffic will be video content by 2014 [47] which will provide a source for large revenue gains. With this in mind it is clear that there is a real need for intelligent bandwidth provisioning.

This thesis presents an experimental analysis by utilising real wireless networks and real video streaming without using simulation packages. The EDCA settings are modified adaptively in real time and the quality of the video stream is objectively analysed using statistical methods. To the best of our knowledge this is the only body of work where this particular approach has been employed.

6.2 Further Work

This work aims to determine the optimal value of *Cmin* to allocate to the video AC in order to satisfy end user quality requirements. In this case, this task was accomplished by means of experimental analysis. Future areas of interest should endeavour to develop a system whereby the *Cmin* required can be calculated by analysing the video file at the host side and taking into account the current network conditions and then supplying that value to the WRRC software. This approach could then be further expanded upon to deliver dynamic *Cmin* tuning based on the user quality requirement and the network conditions.

Another area that would be worthwhile investigating would be to design a control algorithm that modified the EDCA settings based on the received QoE of the video stream. The work in this thesis modifies the EDCA settings based on a user supplied *Cmin* value which then affects QoE. If the EDCA was dependant on QoE then the need to specify the *Cmin* component could be removed entirely. This presents a significant task as there would need to be a feedback channel from the video player at the client side to the access point. This would essentially implement a dynamic *Cmin* based on QoE. This approach could be further extended to modify the *Cmin* setting based on the line rate at which the QAP is transmitting. This approach could then satisfy user QoE with varying video content and client mobility.

The Video server used for these experiments was the Darwin Streaming Server. It would be a worthwhile exercise to utilise another server to determine to what extent the results gathered here would be replicated. Do other servers introduce a distribution of PSNR values as observed with the Darwin streaming server utilised here? In the work detailed within this thesis, even under the best case scenario, the frames arriving at the client had a reduced PSNR. It is possible that if a server delivers all the frames without any drop in PSNR the overall quality could be improved upon.

It would also be beneficial to repeat these experiments with a modified control algorithm built into the CNRI WRRC. Currently the drop in QoE at the higher values of *Cmin* is attributed to the operation of the control algorithm, with this modified the results may vary. The WRRC also needs to be modified in order to take account of mobile nodes. In these experiments the client stations were maintained at a fixed distance from the QAP. If mobility is introduced the *Cmin* values would need to be changed in order to take account of line rate adaptation that may be employed by some QAPs.

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APPENDIX

[AP1] PERL script to batch convert mp4 files to yuv files using ffmpeg.

```
#!/usr/bin/PERL -w

my @file_name;

opendir(DIR, ".");
@files = grep(/\.mp4$/,readdir(DIR));
closedir(DIR);

foreach $file (@files) {

        @file_name = split(/\./, $file);
  print "$file\n";

  print "$file_name[0]\n";

  $file_name[0] = $file_name[0] . ".yuv";

  print "$file_name[0]\n";

  @args = ("ffmpeg.exe", "-i", "$file", "$file_name[0]");

  system(@args);
}
```

[AP2] Matlab program to calculate and plot PDF, CDF and CCDF information based on excel file of PSNR values.

```
function y = process(filename, k)
global colours;
global file_array;
colours = struct('colour', {'b','g','r','c','m','y','k','g:','r-.','c--'});

%file_array = struct('name', {} );

%file_array(k).name = filename;

x = xlsread(filename);
file_out = [filename '.jpg'];

data = x(:,2);
for i = 1 : 1 : max(size(data))
   if data(i) > 100
      data(i) = 100;
   end
end

[fy, xy]=ksdensity(data);


figure(1);     %used to display all pdf curves on one figure
hold on;
plot(xy,fy, colours(k).colour);
title('All PSNR PDF Data');
%legend(file_array.name);
print( '-f1', '-djpeg90', 'all_psnr_pdf_curves.jpg');

figure(11);     %displays current PSNR PDF only
plot(xy,fy, colours(k).colour);
title(file_out);
%legend(filename);
print( '-f11', '-djpeg90', ['.\PDF_PSNR\' file_out]);
close(11);



for i = 1:1:max(size(data))
   if( (data(i) >= 9.3) && (data(i) <= 45) )
      VQM(i) =( (0.0007816 * data(i)^2) - (0.06953 * data(i)) + 1.5789);
   elseif( data(i) < 9.3)
      VQM(i) = 1;
```

```
    elseif (data(i) > 45)
        VQM(i) = 0.033;
    else
    end
end
VQM = VQM';

figure(2);      %displays all VQM PDFs on one figure
hold on;
[fy, xy]=ksdensity(VQM);
plot(xy,fy, colours(k).colour);
title('All VQM PDF Data');
print( '-f2', '-djpeg90', 'all_vqm_pdf_curves.jpg');

figure(22);     %displays current VQM PDF only
plot(xy,fy, colours(k).colour);
file_out = ['VQM-' filename '.jpg'];
title(file_out);
print( '-f22', '-djpeg90', ['.\PDF_VQM\' file_out]);
close(22);



figure(3);      %displays all PSNR CCDFs on one figure
hold on;
[H,STATS] = cdfplotm(data, colours(k).colour);
title('All PSNR CCDF Curves');
print( '-f3', '-djpeg90', 'all_psnr_ccdf_curves.jpg');

figure(33);     %displays current PSNR CCDF only
[H,STATS] = cdfplotm(data, colours(k).colour);
file_out = ['CCDF-' filename '.jpg'];
title(file_out);
print( '-f33', '-djpeg90',  ['.\CCDF_PSNR\' file_out]);
close(33);

figure(4);      %displays all VQM CDFs on one figure
hold on;
[H,STATS] = cdfplot(VQM, colours(k).colour);
title('All VQM CDF Curves');
print( '-f4', '-djpeg90', 'all_vqm_cdf_curves.jpg');

figure(44);     %displays current VQM CDF only
[H,STATS] = cdfplot(VQM, colours(k).colour);
file_out = ['CDF-VQM-' filename '.jpg'];
title(file_out);
print( '-f44', '-djpeg90', ['.\CDF_VQM\' file_out]);
close(44);
```