A Novel QoS-Aware MPEG-4 Video Delivery Algorithm over the Lossy IEEE 802.11 WLANs to Improve the Video Quality

by

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Abstract

Video traffic is bursty in nature and has different network requirements compared to other types of traffic (e.g. voice, data) in terms of bandwidth, delay, jitter, and loss etc. So it becomes important to manage video traffic on a WLAN carefully to achieve acceptable levels of Quality of Service (QoS). The unique contribution of this work is that it presents experimental and simulation studies of the performance of real video content streamed over WLAN networks. Under various test scenarios the performance of the WLAN network in terms of delay, loss, throughput etc. is analysed in the presence of background traffic. The effects of different types of server configurations and access contention between stations are also investigated for IEEE 802.11b and IEEE 802.11e networks. This work specifically considers the *IPB* fame based nature of MPEG-4 encoded video. A novel QoS aware MPEG-4 video delivery algorithm is proposed and evaluated using a computer model written in the *C* programming language. The model exploits two mechanisms namely frame retransmission (ReTx) and GOP truncation (GOPT). The ReTx mechanism effectively increases the QoS by minimising the transmission losses at the expense of an increased buffer overflow probability. The GOPT mechanism reduces the probability of buffer overflow at the expense of a reduced QoS. The QoS aware MPEG-4 video delivery algorithm aims to achieve an optimal trade off between these two mechanisms in order to eliminate buffer overflow and minimise transmission losses. The algorithm aims to replace uncontrolled packet loss due to buffer overflow, MAC collisions, and transmission errors by a controlled prioritized packet loss scheme that permits a graceful degradation in MPEG-4 video quality streamed over IEEE 802.11b networks. This ensures the realisation of the most favourable network conditions for the delivery of MPEG-4 video frames on WLANs. Through extensive simulations it has been shown to provide a significant improvement in the QoS performance for video streaming applications for both uplink and downlink network scenarios in the presence of background traffic.

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Abbreviations and Acronyms

AC	Access Category
AAC	Advanced Audio Codec
ACK	Acknowledgment
AIFS	Arbitration Inter-Frame Space
AIFSN	Arbitration Inter-Frame Space Number
AIPD	Additive Increase Proportional Decrease
AP	Access Point
ATSC	Advanced Television Systems Committee
AVI	Audio Video Interleave
AVC	Advanced Video Coding
BC	Backoff Counter
BER	Bit Error Ratio
BPSK	Binary Phase Shift Keying
BS	Base Station
BSS	Basic Service Set
CBR	Constant Bit Rate
CCK	Complementary Code Keying
CDF	Cumulative Distribution Function
CFP	Contention-Free Period
CIF	Common Intermediate Format
CP	Contention Period
CRC	Cyclic Redundancy Check

CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
CTS	Clear To Send
CW	Contention Window
CW_{min}	Minimum Contention Window
CW_{max}	Maximum Contention Window
dB	Decibel
DBPSK	Differential Binary Phase Shift Keying
DCF	Distributed Coordination Function
DF	Distribution Function
DHCP	Dynamic Host Configuration Protocol
DIFS	DCF Inter Frame Space
DLL	Downlink Load
DSCP	Differentiated Services Code Point
DSSS	Direct-Sequence Spread Spectrum
DVB	Digital Video Broadcasting
ECW_{min}	Exponent form of CW_{min}
ECW_{max}	Exponent form of CW_{max}
EDCA	Enhanced Distributed Channel Access
EIFS	Extended Inter-Frame Space.
ETSI	European Telecommunications Standards Institute
ESS	Extended Service Set
ESSID	Extended Service Set Identification

FCS	Frame Check Sequence
FHSS	Frequency Hopping Spread Spectrum
FIFO	First In First Out
FTD	Frame Transmission Delay
FTP	File Transfer Protocol
GOP	Group of Pictures
GOPT	GOP Truncation
HCF	Hybrid Coordination Function
HCCA	HCF Controlled Channel Access
HDTV	High Definition Television
HR-DSSS	High Rate - Direct Sequence Spread Spectrum
HTTP	Hypertext Transfer Protocol
IBSS	Independent Basic Service Set
IEEE	Institute of Electrical and Electronic Engineers
I, P, B	Intra, Predicted, Bi-directional Frames
IFS	Inter Frame Space
IPD	Inter-Packet Delay
IPRT	Inter-Packet Receiving Time
IPST	Inter-Packet Sending Time
IPTV	IP Television
ISM	Industrial, Scientific and Medical
ISDB	Integrated Services Digital Broadcasting
ITU	International Telecommunications Union

JPEG	Joint Photographic Expert Group
LAN	Local Area Networks
LLC	Logical Link Control
MAC	Medium Access Control
MIMO	Multiple Input Multiple Output
MPDU	MAC Protocol Data Unit
MPEG	Moving Pictures Expert Group
MSDU	MAC Service Data Unit
MSE	Mean Square Error
MTU	Maximum Transmission Unit
NAV	Network Allocation Vector
NTP	Network Time Protocol
NTSC	National Television System Committee
OFDM	Orthogonal Frequency-Division Multiplexing
OSI	Open System Interconnection
PAL	Phase Alternating Line
PC	Point Coordinator
PCF	Point Coordinator Function
PCMCIA	Personal Computer Memory Card International Association
PDU	Protocol Data Unit.
PESQ	Perceptual Evaluation of Speech Quality
PEVQ	Perceptual Evaluation of Video Quality
PHY	Physical Layer

PIFS	PCF Inter Frame Space
PLCP	Physical Layer Convergence Procedure
PMD	Physical Medium Dependent
PMR	Peak to Mean Ratio
PPDU	PLCP/ Physical (layer) Protocol Data Unit
PPS	Packets Per Second
PSDU	PHY/PLCP Service Data Unit – PLCP SDU
PSQM	Perceptual Speech Quality Measurement
PSNR	Peak Signal to Noise Ratio
QAP	QoS Aware Access Point
QCIF	Quarter Common Intermediate Format
QPSK	Quadrature Phase Shift Keying
QoD	Quality of Delivery
QoE	Quality of Experience
QoS	Quality of Service
QSTA	QoS Aware Station
RETX	Re - Transmission
RSSI	Received Signal Strength Indicator
RTP	Real Time Protocol
RTS	Request To Send
RTCP	Real Time Control Protocol
RTSP	Real Time Streaming Protocol
SAP	Service Access Point

SECAM	Séquentiel Couleur Avec Mémoire, French for "Sequential Color with Memory"
SDTV	Standard Definition Television
SIFS	Short Inter Frame Space
SNAP	Sub-Network Access Protocol
SNR	Signal-Noise Ratio
SSID	Service Set Identity
STA	Wireless Station
TCP	Transmission Control Protocol
TSPEC	Traffic Specification
TXOP	Transmission Opportunity
UDP	User Datagram Protocol
ULL	Uplink Load
VBR	Variable Bit Rate
VoIP	Voice over Internet Protocol
VQM	Video Quality Measurement
WEP	Wired Equivalent Privacy
Wi-Fi	Wireless Fidelity
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Network
WMM	Wi-Fi Multimedia
WPA	Wi-Fi Protected Access
WPAN	Wireless Personal Area Network

Chapter 1 INTRODUCTION

Wireless communications has enjoyed spectacular growth during the last decade as traditional means of wired communications have proven to be inadequate in meeting the ever-changing requirements of users. Setting up a wired network is relatively expensive in terms of cost, labour, and time. As the number of users and their requirements such as bandwidth, speed, desire for multimedia services etc. are ever growing; it is a cumbersome task to continually upgrade the network. Wireless networks offer many advantages over traditional wired networks. Wireless technologies can be an alternative solution in situations where network cabling is difficult or not feasible (e.g. historic or protected buildings, battlefields, remote areas, areas hit by natural disasters etc.). Other benefits include ease of deployment, simplicity, greater flexibility, reduction in infrastructure and operating costs etc. Currently a diverse range of wireless technologies are deployed around the globe.

Some of the most popular wireless technologies include the mobile or cellular communication systems [1], Bluetooth [2], WiMax [3,4], and Wireless Local Area Networks (WLANs) [5]. The WLAN is a wireless extension of the traditional wired LANs and allows for two or more devices to communicate without network cabling using standard network protocols. WLANs transmit and receive data over the air using electromagnetic waves, eliminating the need for wired connections. The IEEE 802.11 standard was developed by the IEEE LAN/MAN Standards Committee (IEEE 802) for WLAN. IEEE 802.11 has many family members with IEEE 802.11 a/b/g/n being the most popular.

WLANs range of operation is typically less than 100 metres which is sufficient for small to medium enterprises and residential houses. As WLANs operate in the unlicensed ISM bands, there can be other devices which use the same frequency band and may lead to signal interference [6]. Originally, the IEEE 802.11 WLANs provided a best-effort service only, i.e. it does not differentiate between data and real time traffic such as voice and video. It gives all traffic types the same priority. With the introduction of the IEEE 802.11e standard, it is possible to provide a prioritised service to real time traffic, for example VoIP and video conferencing services, in order to ensure higher throughput with low delay, loss and jitter.

1.1 Framework of the Thesis and Motivation

Streaming video over networks is an important and active area of research. Research areas include applications, content, encoding, server transmission, adaptation, client, quality assessment, quality improvement etc. Video streaming can be defined as a server/client technology and can be delivered by either peer-to-peer (unicast) or broadcast (multicast). The main goal of streaming is that the video packets should arrive and play out continuously with as small a delay and loss as possible to achieve acceptable levels of Quality of Service (QoS) within the constraints of the bandwidth available. QoS is the term often used to describe the overall quality of a video. QoS is actually composed of two separate elements [7] - Quality of Delivery (QoD) and Quality of Experience (QoE). QoD describes how a stream is affected by network conditions such as packet loss, delay, jitter etc. QoE [8,9] relates to how an end user perceives the visual quality of the played out video. This thesis will only consider the QoD aspect of video streaming over WLANs. There are different types of video standards (e.g. MPEG -1/2/4, H .263, H .264 etc.) and WLANs (e.g. b/g/a/n) available. It is extremely difficult to propose a solution that would suit all WLANs and video standards. The goal of this research was to propose and implement an *Adaptive Video Streaming Scheme* which would be generic in nature for different types of IEEE 802.11 WLANs and video standards which use the *IPB* frame hierarchy levels. Although this work describes the streaming of MPEG-4 video over IEEE 802.11b WLANs, the proposed solution would work with other WLAN standards which implement MAC buffers and *IPB* frame based video standards. The validated proposal can guarantee a significant QoS performance improvement for video applications over WLANs.

1.1.1 Problem Statement

WLANs pose a significant challenge for delivering video streaming services as they have lower data rates and higher error rates compared to wired networks. Currently traditional IEEE 802.3 or Ethernet/ Wired Local Area Networks (LANs) can reliably offer data rates of up to 1 Gbps where as WLANs may achieve data rates in Gbps range within the next 2 to 3 years. WLANs are also a best effort data service. The capacity is not fixed and depends on the nature of the traffic load. Consequently, streaming video may not be allocated sufficient bandwidth to be streamed with an acceptable QoS over WLAN networks. Video traffic is also bursty in nature and tends to be characterised by large packet sizes and hence large bandwidth requirements. Due to the nature of video traffic and the hostile nature of the wireless environment, WLANs are not ideally suited for delivering video with an acceptable QoS. Consequently, streamed video traffic requires a different treatment from other traffic on the WLAN. There are different metrics available to evaluate the quality of the streamed MPEG-4 video over WLANs such as delay, loss, jitter, throughput etc.

WLANs employ a MAC mechanism which is contention based, i.e. users need to contend for access to the medium and the medium is shared between all users. Increased contention for access leads to an increase in the time required to win a transmission opportunity which increases the delay time of a packet in the transmit buffer awaiting transmission. An increased contention also leads to an increased probability of collision which in turn requires a greater number of retransmissions to minimise the packet loss. A consequence of retransmissions is that the delay time required to successfully transmit a packet increases. Any increase in the waiting time of a packet in the transmit buffer leads to an increase in the probability of buffer overflow as the transmit buffer is filled up at a shorter period of time. So it becomes important to manage the transmission of the video traffic on a WLAN carefully.

Video frames can get lost on a WLAN in different ways. These are – transmission errors due to noise and interference present in the medium, buffer overflow due to an insufficient availability of transmission opportunities to satisfy the incoming video frames and MAC collisions arising from contention for access. IEEE 802.11 WLANs operate in unlicensed public bands, hence interference from other devices operating in the same frequency bands, e.g. Bluetooth, cordless phones, wireless cameras etc. is a reality. Interference can have a negative impact on the performance of video streaming by increasing the probability of packet losses in the medium. In this work only loss metric (Mac collision and buffer overflow loss) has been considered.

1.1.2 Significance of the Problem

According to a recent Cisco study [10] studies video traffic on global IP networks is expected to account for 90% of all Internet traffic by the end of 2012. Another study [11] conducted by the International Data Corporation (IDC) provides a five-year forecast for 2011–2015 for the online video platform market with the conclusion that video market will grow rapidly over the next couple of years to more than $1 billion in 2015. In other words, video will soon become the dominant traffic over IP networks. In a typical wireless network various types of traffic (e.g. video, voice, data etc.) may be present. Video traffic quickly exposes any weaknesses in the network. The available bandwidth or capacity of a typical WLAN is finite; moreover it is not fixed and is load dependent. Streamed video traffic has different network requirements compared to other types of traffics in terms of bandwidth, delay, jitter, and loss. In order to deliver video with an acceptable QoS there are certain criteria (as minimum bandwidth, maximum delay and loss rate) which have to be satisfied. The performance of video services are quite sensitive as uncontrolled packet loss due to buffer overflow, MAC collisions and transmission losses contribute to screen freeze, and audio quality distortion and the viewer’s experience would be unsatisfactory. Also, due to the hostile nature of the WLAN environment, video frames can get lost while being transmitted on the medium thus drastically reducing the video quality in an uncontrollable fashion. Packets lost due to collisions can be retransmitted but at the expense of a higher buffer overflow probability. Buffer overflow occurs when there is insufficient capacity in the transmit buffer to accommodate the arrival of new packets to be transmitted. This can lead to a catastrophic drop in the QoS since the packets lost due to buffer overflow cannot be recovered. To deliver video packets with acceptable QoS uncontrolled packet losses should be minimised and packets dropped at the MAC layer due to buffer overflow need to be eliminated. Hence it is quite challenging to guarantee acceptable QoS for streamed video over WLANs.

1.1.3 Contributions of This Thesis

To design a video over WLAN system, some important issues need to be considered so as to guarantee the performance of the network. From a network engineer's perspective, bandwidth and QoS (which includes delay, loss etc.) are among the most important issues.

Experimental results (described in detail in appendix section) suggest that by exploiting the *IPB* frame based nature of MPEG-4 video the QoS of the streamed video can be improved. Based on this finding, this thesis proposes a novel QoS aware MPEG-4 video delivery algorithm (described in chapter 4 and validated in chapter 5) which employs two mechanisms namely frame retransmission (ReTx) and GOP truncation (GOPT). The first mechanism (i.e. ReTx) is well known and is focused on minimizing packet loss due to MAC collisions and transmission impairments. The novel GOPT mechanism proposed here involves selectively dropping frame triplets from the GOP to reduce the number of video packets required to be transmitted. A Group of Pictures (GOP) size is defined as the length between two successive *I* frames. MPEG-4 standard defines a 15 frame GOP with the following frame sequence - *IBB PBB PBB PBB PBB.*

The proposed QoS aware MPEG-4 video delivery algorithm is based on the measurement of the buffer occupancy metric which is measurable through implementing the MAC buffers for the IEEE 802.11 WLANs. The ReTx and GOPT mechanisms are applied successively. It achieves the ITU-T target specified for loss rate (< 1%) of streamed video transmission. This ensures the realisation of the most favourable network conditions for the delivery of MPEG-4 video frames on WLANs. The algorithm aims to replace uncontrolled packet loss due to buffer overflow and MAC collisions by a controlled prioritized packet loss scheme that permits a graceful degradation in QoD for MPEG-4 video streamed over IEEE 802.11b networks.

In particular a trade-off exists between the number of retransmitted frames due to frame loss and number of frames present in the GOPs. The more the ReTx mechanism is being used the more the bandwidth is required. In order to reduce the extra bandwidth required for retransmitting the frames lost due to MAC collisions, GOPT is being applied. ReTx trades off bandwidth for QoS and GOPT mechanism trades off QoS for bandwidth. The optimal trade-off between the two mechanisms will be determined by network and traffic conditions such as video content, capacity of the system, contention present in the medium, packet size, packet rate etc. with the target of avoiding buffer overflow (BO) and minimising transmission losses. Hence, the probability of BO will govern the choice of optimal GOPT and ReTx levels so that the system can ensure maximum QoS under the given operating conditions by aiming to replace uncontrolled frame loss by controlled or prioritized frame loss.

The study of the proposed QoS aware MPEG-4 video delivery algorithm is based upon computer simulation using computer models developed in the *C* programming language. The QoS aware MPEG-4 video delivery algorithm has been validated in chapter 5 through extensive simulations for both uplink and downlink video traffics over the IEEE 802.11b WLANs in the presence of the CBR background traffic with the goal of optimising quality of streamed MPEG-4 video traffic. Various modelling parameters have been extracted from twelve different real life MPEG- 4 video clips (of six genres: Computer Generated Imagery– CGI, Action, Animation, Sport, Documentary, and Talking Head) which were subsequently used in validating the proposed novel algorithm to analyse the QoS of the streamed video. The video clips are of five minutes duration and each taken from different movies and other sources as described in detail in chapter 5.

It has been shown to provide a significant improvement in the QoS performance for video streaming applications for both uplink and downlink network scenarios in the presence of background traffic through extensive simulations. In the uplink scenario, it was observed that the network's video capacity is 2 streams. Afterwards the maximum tolerable background loads for 1 and 2 video streams were obtained which were 2.4 and 0.5 Mbps respectively. It was demonstrated that to obtain zero buffer occupancy after employing frame ReTx, GOPTs in the region of 20% to 60% were required for different video clips. In the downlink case it was observed that when the QoS aware MPEG-4 video delivery algorithm was not implemented, for all twelve video clips there was an average ~5% frame loss for all three frame types. This percentage of frame loss translated into ~10% -18% loss in bandwidth. However, when the ReTx mechanism was applied the frame loss rate reduced to the target $\leq$ 1% level which means that more frames could be delivered successfully and consequently the net savings in bandwidth were observed in the range of ~9% -17%.

In summary, this thesis proposes an alternative use of the ReTx mechanism provided for under the IEEE 802.11b standard by exploiting the frame based nature of MPEG-4 videos by proposing a novel QoS aware MPEG-4 video delivery algorithm. The algorithm advocates the combined use of GOPT in frame triplets (*PBB*) and ReTx to minimise the probability of uncontrolled packet loss of the video streams at the expense of reduced quality thus achieving controlled and graceful video quality degradation under heavy network loads. Hence the algorithm aims to deliver a QoS improvement by ensuring the realisation of the most favourable network conditions for the delivery of MPEG-4 video frames on WLANs. It would work with all types of IEEE 802.11 based WLANs (e.g. b/g/a/n) although it is proposed for IEEE 802.11b WLANs only due to its generic nature. It is also applicable for a wide range of video contents (e.g. H.263/.264) other than the MPEG-4 format which can be segregated into their constituent *IPB* frames as it is concerned with buffer occupancy.

1.2 Thesis Outline

This thesis is organised as follows:

Chapter 2 details the technical background of this work. This thesis specifically deals with MPEG-4 video streaming over WLANs. Related IEEE 802.11 standards, video characteristics and video streaming concepts (e.g. QoS) are detailed here.

Chapter 3 presents a thorough literature review of relevant and up to date work to highlight the recent advances in this active research field and how it applies to this work.

Chapter 4 describes the experimental tools used and analyses the experimental results obtained.

Based on the knowledge gained from chapter 4 regarding the influence of the *IPB* frame based nature of the MPEG-4 video, chapter 5 presents the QoS aware MPEG-4 video delivery algorithm developed to replace uncontrolled video QoS degradation with controlled graceful QoS degradation. The algorithm has been validated through extensive simulation for both uplink and downlink video traffics.

Chapter 6 details the summary of this research work, i.e. main findings with concluding remarks. Suggested future directions are also provided with regard to further enhancing video delivery over WLANs.

1.3 Publications Arising from This Work

1. Cranley N, **Debnath T.**, Davis M. "*Video Streaming Applications over WLAN***"**, Poster: NCNRC, Maynooth, Ireland, March 2006.

2. **Debnath T**., Cranley N., Davis M. "*Experimental Comparison of Wired versus Wireless Video Streaming over IEEE 802.11b WLANs*", in Irish Signals and Systems Conference (ISSC '06), Dublin, Ireland, June 2006.

3. **Debnath T.**, Cranley N., Davis M. "*Experimental Investigation of the Effects of Background Traffic Loads on Streamed Video over 802.11b WLANs*", in the Information Technology & Telecommunications Conference (ITT '06), Carlow, Ireland, October 2006.

4. Cranley N., Debnath T., Davis M. "*An Experimental Investigation of Parallel Multimedia Streams over IEEE 802.11e WLAN Networks using TXOP",* in the IEEE International Conference on Communications (ICC '07), Glasgow, The United Kingdom, June 2007.

5. Cranley N., **Debnath T.**, Davis M. "*The Effects of Contention between stations Video Streaming Applications over Wireless Local Area Networks- an experimental approach*", in the Information Technology & Telecommunications Conference (ITT '07), Blanchardstown, Ireland, October 2007.

Chapter 2 TECHNICAL BACKGROUND

2.1 Introduction to Wireless Local Area Networks

Through evolution the IEEE 802.3 (Ethernet) and IEEE 802.11 (WLANs) standards have endured and have become the dominant networking standards over the years for wired and wireless communications respectively. As of today, WLAN is the most widely deployed wireless technology [12]. The IEEE 802.11 standard was developed by the IEEE LAN/MAN Standards Committee (IEEE 802) for WLAN. IEEE 802.11 has many family members with IEEE 802.11 a/b/g/n being the most popular. The IEEE 802.11e is an enhancement to the original IEEE 802.11 standards which specifically addresses QoS for real-time applications such as voice and audio. The first version of the IEEE 802.11 standard was ratified in 1997 [13]. Then IEEE 802.11b and IEEE 802.11a were both standardized in September 1999. The IEEE 802.11b boosted the line rate of IEEE 802.11 from the original 1 or 2 Mbps rate to 11 Mbps. The IEEE 802.11a increased that to 54 Mbps by using the 5 GHz frequency band. The IEEE 802.11g standard was approved in June of 2003, which works at the 2.4 GHz band and at maximum 54 Mbps rate. In September of 2009, the IEEE 802.11n standard was ratified which allows for a throughput in excess of 100 Mbps in both the 2.4 GHz and 5 GHz bands using channel bonding with up to 72 Mbps without channel bonding.

2.1.1 Different Standards

The different IEEE 802.11 working groups are

- 802.11a (which supports rates of up to 54 Mbps in the 5 GHz ISM band)
- 802.11b (which supports rates of up to 11 Mbps in the 2.4 GHz ISM band)
- 802.11c (Wireless AP Bridge Operations)
- 802.11d (Internationalization)
- 802.11e (which defines QoS enhancement mechanisms)
- 802.11f (which addresses the interoperability of APs /stations from different vendors)
- 802.11h (supports power control for 5 Ghz range- requirement for operation in Europe)
- 802.11g (which supports rates of up to 54 Mbps in the 2.4 GHz ISM band)
- 802.11i (which deals with security issues)
- 802.11n (implements high data rates > 100 Mbps,)
- 802.11p (wireless access for vehicular environments)
- 802.11s (Mesh networking)

The IEEE 802.11 a/b/g standards support best effort services only, whereas the IEEE 802.11e standard provides for service differentiation mechanisms. In the IEEE 802.11a/b/g, standards, the emphasis is on enhancements to the PHY layer while for IEEE 802.11e the focus is shifted to enhancements of the MAC sub-layer.

IEEE 802.11b

This was ratified in 1999 and supports transmission speeds of up to 11 Mbps and operates in the 2.4 GHz ISM (Industrial, Scientific, and Medical) band. IEEE 802.11b uses DSSS (Direct Sequence Spread Spectrum) with a single carrier per channel. There are four possible transmission rates defined, i.e. 1, 2, 5.5 and 11 Mbps [14].

IEEE 802.11a

This operates in the 5 GHz ISM band with transmission speeds of up to 54 Mbps. There are 8 rates defined, i.e. 6, 9, 12, 18, 24, 36 48 and 54 Mbps. But only 6, 12, and 24 Mbps are mandatory with the rest being optional. IEEE 802.11a uses OFDM (Orthogonal Frequency Division Multiplexing) which is a form of FDMA where the data stream is divided into several lower-rate streams which then are transmitted simultaneously on multiple sub-carriers. The sub-carriers are modulated using BPSK, QPSK, 16-QAM or 64-QAM modulation [15].

IEEE 802.11g

This uses the same modulation technique as IEEE 802.11a, but in the 2.4 GHz ISM band. IEEE 802.11g supports transmission speeds up to 54 Mbps using OFDM modulation. It was ratified in June 2003. It also incorporates mechanisms to ensure backward compatibility with existing IEEE 802.11b systems [16].

IEEE 802.11e

This standard was ratified in late 2005 and defines a MAC enhancement to the original IEEE 802.11 to address the QoS issues for the delivery of voice and video services over WLAN. This will work on all physical layer specifications defined in IEEE 802.11. The IEEE 802.11e defines a series of QoS enabling mechanisms. QoS is supported through the creation of four separate queues. Each queue (known as Access Category) has four different configurable parameters namely *AIFSN*, $CW_{\min}$, $CW_{\max}$, and *TXOP* [17].

IEEE 802.11h

This standard includes transmission power control and dynamic frequency selection to reduce interference and comply with European regulations in the 5 GHz band [18].

IEEE 802.11n

This standard was ratified in September, 2009 and aims to achieve much higher data rates (>100 Mbps) than previous IEEE 802.11 standards by modifying both the PHY and MAC sub-layers using MIMO technology in both the 2.4 GHz and 5 GHz bands [19,20,21]. This relatively new standard intends to improve the QoS of streaming multimedia by essentially *throwing bandwidth* at the problem. It claims to support transmission speeds up to 150 Mbps per stream and up to four streams, can be up to 12 times faster than current IEEE 802.11a and IEEE 802.11g technology, and it uses greater efficiency to deliver up to 20 times the throughput of legacy standards. Depending on the enterprise goals adopting this standard demands trade-offs between range and performance to address user density and bandwidth considerations. Sites that require maximum coverage generally exhibit low user density and throughput demands. Supporting a small number of low traffic Wi-Fi client devices scattered over a large area, these sites require only a few access points to provide adequate wireless service. On the other hand, sites that require maximum capacity need to serve many concurrent users with high bandwidth requirements, e.g. real-time applications such as voice, video and location tracking. This standard will be discussed in greater detail at a later section.

IEEE 802.11ac and 802.11ad

There are two yet-to-be-approved (as of August 2011) WLAN standards - 802.11ac and 802.11ad. The goal of IEEE 802.11ac is to provide data speeds of around 1 Gbps. It is expected that a draft standard would be available during late 2011/early 2012 and products out by the end of 2012. The technology that IEEE 802.11ac will use to achieve a high data rate of 1 Gbps has not been finalized. It may involve using wider channels, bonding four to eight channels together, and implementing some high level engineering to the modulation scheme involved. The IEEE 802.11ad standard will use the 60 GHz band to provide fast throughput. Due to high frequency and limited penetration through walls, the unlicensed 60 GHz band is relatively quiet and noise-free. The idea is to produce a standard which would switch to 60 GHz when a high-speed, short-range transmission is required, but would fall back to conventional Wi-Fi (i.e. IEEE 802.11 b/a/g/n) using 2.4 GHz or 5 GHz at other times [22].

2.1.2 General Description of the IEEE 802.11 WLANs

The main components of an IEEE 802.11 WLAN system are

1. Stations (STAs)
2. Access Point (AP)
3. Basic Service Set (BSS)
4. Extended Service Set (ESS)
5. Distribution System (DS)

Stations (STAs)

These are the devices containing a network interface card that connect to the wireless medium. Stations contain IEEE 802.11 MAC and PHY layers and support station services such as authentication, de-authentication, privacy, reliable delivery of data from MAC of one station to MAC of other stations, etc.

Access Point (AP)

This is the central base station or bridge between the wireless and wired networks. In order to find networks to connect to, stations scan for active networks announced by access points. Before sending data, stations must associate with an access point. An AP provides distribution system services such as association, disassociation, reassociation etc. An AP has a finite operational range within which a wireless connection can be maintained between the client station and the access point which is typically 50-150 metres with performance degrading with distance. The actual distance depends on many factors, e.g. the propagation environment, building construction etc.

Basic Service Set (BSS)

This is the set of stations that communicate with each other in a basic building block. There are two types of BSS: Independent BSS (IBSS)/Ad hoc mode and Infrastructure BSS (BSS).

The difference between these two types is determined by the presence or absence of an AP. When there is no AP present, the network is defined to be operating in the Ad hoc mode. The infrastructure mode includes an AP. All stations communicate directly with the AP. The AP provides a connection to the wired LAN and also provides relay functionality. The AP provides for centralised control of the BSS.

Extended Service Set (ESS)

ESS is a set of infrastructure BSSs. Here APs communicate with each other and traffic is forwarded from one BSS to another. This system facilitates the movement of stations from one BSS to another. So the range of mobility is extended beyond the reach of a single BSS. The system has a common distribution system (DS) and the same SSID is shared.

2.2 IEEE 802.11 Architecture

Architectures and protocols define how a particular LAN operates. Protocols are set of rules that govern communication between peer entities or networks. WLAN and other LAN standards (e.g. Ethernet IEEE 802.3, Token Ring- IEEE 802.5) are compatible above the data link layer (DLL). The DLL and PHY layers are different in IEEE 802.11.

For LAN implementation, the Data Link Layer is divided into two sub-layers:

- Logical Link Control (LLC)
- Medium Access Control (MAC)

The LLC is standardised in IEEE 802.3 or Ethernet. It deals with interfacing to higher levels and flow and error control. The Network Layer on sender passes a packet to the LLC, using LLC access primitives. The LLC sub-layer then adds a LLC header, containing sequence and acknowledgement numbers. The resulting structure is then inserted into the payload field of an IEEE 802.11 frame and transmitted. At the receiver, the reverse process takes place.

The MAC sub-layer is primarily responsible for controlling access to the wireless medium. The main mode for accessing the network medium is a traditional contention-based access method, though it employs collision avoidance (Carrier Sense Multiple Access with Collision Avoidance- CSMA/CA) with binary exponential back-off rather than collision detection (CSMA/CD) as used by the IEEE 802.3 Ethernet standard. Other MAC services include authentication, privacy, association, re-association, and power management.

2.2.1 CSMA with Collision Detection (CSMA/CD)

A sending station must sense the carrier, i.e. it has to listen for a clear medium first before transmitting. The word “carrier” in this sense refers to an electrical signal on the cable. If medium is found to be idle, it can proceed with a transmission. It must listen to the network all the time while transmitting. If noise bursts are detected which indicate collision, transmission is aborted. So CSMA/CD with a single channel is inherently a half-duplex system. Valuable resources like time and bandwidth can be saved by terminating damaged frames.

2.2.2 CSMA with Collision Avoidance (CSMA/CA)

Collision detection is rarely performed on wireless networks for engineering reasons. For example, most radios operate in half duplex mode meaning they cannot transmit and receive at the same time on a single frequency. Hence, collision avoidance (CA) is preferred over collision detection (CD). Another reason is due to the large dynamic range (ratio of the largest to smallest signal) of signals on the medium which makes it difficult to distinguish weak incoming signals from noise [23]. The simplest collision avoidance mechanism is to detect or sense the channel to determine whether or not there is a transmission in progress before starting a transmission. This is done in CSMA/CA. So it is essentially a ‘listen before talk’ protocol.

There are two coordination functions through which access to the wireless medium can be controlled. The first is the Distributed Coordination Function (DCF) that uses CSMA/CA and is contention-based; the other is the Point

Coordination Function (PCF) which is based upon polling and is intended for supporting the transmission of real-time traffic. In reality only DCF is implemented as it is mandatory unlike PCF which is optional and has been largely ignored by equipment manufacturers. In this study we will consider DCF only.

2.2.3 Distributed Coordination Function (DCF)

The DCF may be used in either IBSS networks or in infrastructure networks as it allows multiple stations to communicate with each other without a central control. It is considered to be a fair access mechanism, i.e. all contending stations have an equal probability of gaining access to the medium. The DCF mode does not differentiate between STAs, therefore all stations experience the same level of QoS. DCF is intended for best effort traffic and is not suited to real-time traffic such as voice and video. DCF can only support best effort services and cannot give any QoS guarantees, i.e. there is no differentiation mechanism present to guarantee bandwidth, packet delay and jitter for high priority stations or multimedia streaming. Throughput degradation and high delay are caused by the increasing time required for channel access especially under high loads where there is significant competition or contention for access.

2.2.3.1 MAC Frame Types

There are three types of frames defined in the IEEE 802.11 standard - Management, Control, and Data Frames.

Management frames are used for timing, synchronization, authentication, and de-authentication. They are also involved during the association and disassociation of STAs with an AP. Hence they assist in performing the extended operations of the IEEE 802.11 MAC. Common IEEE 802.11 management frames are – authentication, deauthentication, association request, disassociation, beacon, probe response etc.

Control frames are used in conjunction with data frames to perform area clearing operations, channel acquisition and carrier-sensing maintenance functions, and positive acknowledgment of received data. Control and data frames work in conjunction to deliver data reliably from station to station. Examples are Request to Send (RTS), Clear to Send (CTS), and Acknowledgement (ACK) frame. Data frames are responsible for data transfers from station to station.

2.2.3.2 The Access Method

Before attempting to transmit a frame, a station listens to establish that the shared medium is idle. Both physical carrier sensing (performed at the physical layer) and virtual carrier sensing (provided by the network allocation vector at the MAC layer) are performed. If the channel is idle (no frame being transmitted) it may transmit a packet after waiting for a short period of time known as *DIFS* (DCF Inter Frame Space). When the packet reaches the destination, the destination station waits for a time *SIFS* (Short Inter Frame Space) and then it sends an acknowledgment (ACK) frame to the sending station to announce that the transmission was successful. When the medium is busy, all other stations must wait for the channel to become idle. In the meantime all other stations maintain a random back-off interval counter which they start decrementing when the medium is sensed idle, i.e. after the transmission has finished. The backoff counter (BC) is initialised by randomly choosing an integer within a contention window (CW) which is segmented into time-slots. The decrementing of the BC is frozen when the station senses the medium is busy and is resumed when the medium is free for a time period of a *DIFS*. When a station's BC reaches zero, it transmits its packet. When several stations are attempting to transmit, the station that picks the lowest random number wins access to the medium first. If two or more stations transmit at the same time, a collision occurs. The collision is resolved by having the stations involved restart their random access processes again, but with a CW that has been doubled. CW sizes are always 1 less than an integer power of 2 (e.g., 31, 63, 127, 255, 511, and 1023). Each time the retry counter increases, the contention window moves to the next greater power of two. This algorithm is known as the Binary Exponential Back-off Algorithm. By having the randomization interval grow exponentially as more and more consecutive collisions occur, the algorithm ensures a low delay when only a few stations collide but also ensures that the collision is resolved in a reasonable interval when many stations collide. The contention window is reset to its minimum size when frames are transmitted successfully, or the associated retry counter is reached, and the frame is discarded.

In Fig. 2.1, two stations A and B are present. Let's assume, at any instance station B is transmitting. When it is finished transmitting, both the stations pick two randomly generated BC values after a period of *DIFS*.

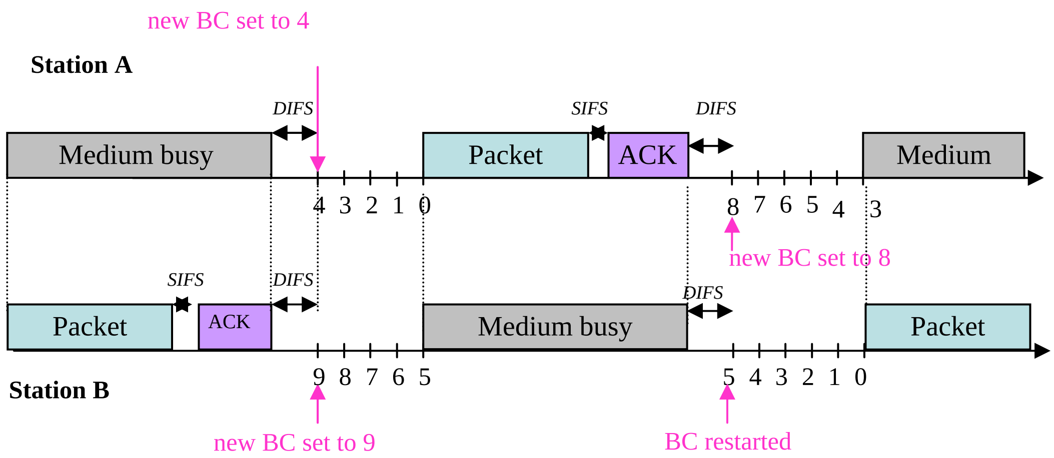

Fig. 2.1: DCF Operation

As station A chooses the lower value it finishes decrementing its BC earlier and hence obtains the right to transmit its frame. As soon as station A starts transmitting, station B stops decrementing its BC. When station A finishes then after a period of *SIFS*, ACK frame transmission time and *DIFS*, it selects another random number. But station B restarts its BC from where it stopped prior to station A's transmission. In this case as station B's BC reaches zero earlier and starts transmitting its frame.

2.2.3.3 Inter-Frame Spacing (*IFS*)

By using different interframe spaces (Fig. 2.2), as well as avoiding collisions, the CSMA/CA mechanism can provide different priority levels for different traffic types. When the medium is idle, high priority traffic can access the medium before low priority traffic by using shorter interframe spaces.

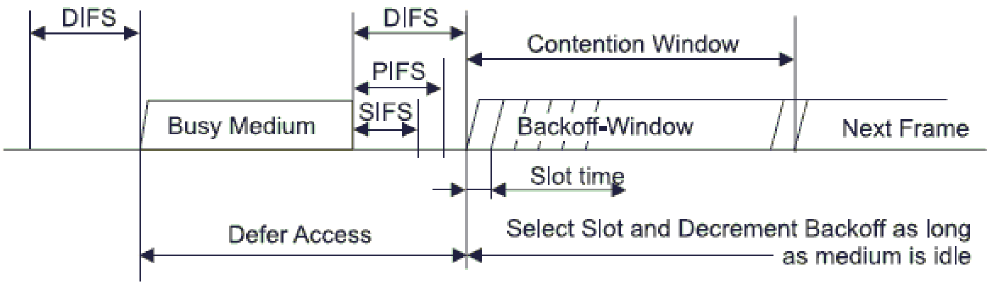

Fig. 2.2: Basic Access Method for a Contending Station.

There are 4 types of *IFS* specified. They are *SIFS*, *DIFS*, *PIFS*, and *EIFS*. The *SIFS* is used for the highest-priority transmissions, such as RTS/CTS frames and ACK frames. The PCF interframe space (*PIFS*) is used by the PCF to provide contention free operation. The *DIFS*, which has the lowest priority, is the minimum idle time a contending station has to wait in order to gain access to the medium. When there is an error in frame transmission *EIFS* (Extended InterFrame Space) is used in DCF mode. *EIFS* is not a fixed interval. *SIFS* is defined by the PHY used. The duration of *SIFS* in IEEE 802.11b and IEEE 802.11a is defined to be 10 and 16 μs respectively. The numerical values of *PIFS* and *DIFS* are calculated according to equations 2.1 and 2.2.

$$PIFS = SIFS + SlotTime \quad (2.1)$$

$$DIFS = SIFS + 2 * SlotTime \quad (2.2)$$

2.2.4 Shortcomings of the IEEE 802.11b Networks

Typically IEEE 802.11b networks operate on a best-effort delivery basis, which means that all traffic is treated with equal priority, i.e. all traffic enjoys the same probability of winning an access opportunity to the medium. The fundamental problem with IEEE 802.11b standard is that it uses a single buffer (shown in Fig. 2.3) for storing the packets while they wait to gain access to medium. Consequently, all packets irrespective of their relative priority are queued in this buffer before being transmitted.

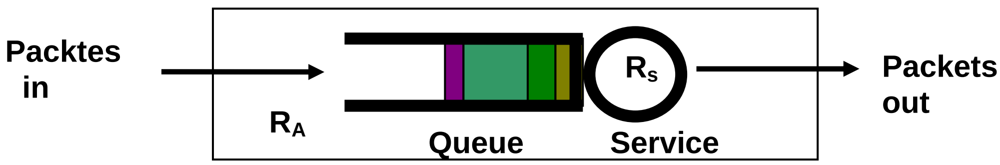

Fig. 2.3: IEEE 802.11b Buffer Queue.

The two factors that determine the performance of a queue are the mean arrival rate (R_A) and the mean service rate (R_S). The average rate at which packets arrive in the queue for service is known as the mean arrival rate. The arrival rate must be greater than or equal to zero. The average time required to service the packets is known as the mean service rate. For a stable system, the mean arrival rate should be less or equal to the mean service rate, i.e.

$$\frac{R_A}{R_S} \leq 1 \quad (2.3)$$

Packets per second (pps) is the common unit for both the rates. The arrivals and service distributions can be one of several types ranging from uniform to random. The queuing discipline describes how the server decides which packet in the queue to pick next for service. Common disciplines are:

- First in first out (FIFO) - in which the packets are processed through the queue in the order in which they are received.
- Last in first out (LIFO) - in which the most recent arrival is served first.

The most common queuing discipline is FIFO (First In First Out).

Queuing Delay is the delay between the point of entry of a packet in the transmit queue to the actual point of transmission. If there are too many packets waiting to be served and the arrival rate is greater than the service rate, the buffer overflows and it starts losing packets.

The size of buffer is also important. IEEE 802.11b standard has a limited buffer space. If the buffer size is relatively large the buffer transit time for a packet will increase but the probability of packet loss will decrease as more packets can be accommodated. On the contrary, if the buffer is relatively small, transit time for a packet will decrease but it would be able to handle a smaller number of packets and the probability of packet loss will increase. Hence, no guarantee of QoS can be ensured for IEEE 802.11b networks.

2.2.5 QoS for IEEE 802.11e – Enhancements to the MAC

The IEEE 802.11b standard provides a best-effort service only which is not suited to real-time video and voice applications. In general video applications require a large bandwidth (typically from 0.5 Mbps to 4 Mbps) to ensure high quality. In order to accommodate these requirements, in May 2000 the IEEE 802.11 Working Group initiated the IEEE 802.11e Task Group to provide support for QoS in delivering real time services. The IEEE 802.11e standard was approved in late 2005 and provides mechanisms to prioritise multimedia traffic over data traffic. Essentially, IEEE 802.11e provides for QoS support through enhancement of the MAC sub-layer. It enables an AP to schedule resources based on client/station data rate and latency needs, improves wireless bandwidth efficiency and packet overheads, and reduces latency by prioritizing wireless packets based on traffic type. Since IEEE 802.11e pertains to the MAC layer, it is compatible with all PHY layers. This enhanced MAC is backwards compatible with original IEEE 802.11 MAC. A station that supports the IEEE 802.11e QoS enhancement mechanisms is referred to as a QoS Enhanced Station (QSTA); whereas an access point that supports these mechanisms is referred to as a QoS Enhanced AP (QAP).

2.2.6 IEEE 802.11e Modes

The IEEE 802.11e standard introduces the Hybrid Coordination Function *(*HCF*)*, which combines functions from DCF and PCF with enhanced QoS-specific mechanisms and frame types. The HCF has two modes of operation—Enhanced Distributed Channel Access (EDCA*)* and HCF Controlled Channel Access (HCCA). EDCA is contention based and is intended for support of differentiated QoS. HCCA works by controlling channel access and is intended for parameterised traffic during contention free periods. In this standard the MAC frame structure is enhanced by adding a new field known as QoS Control as shown in Fig. 2.4.

Bytes 2	2	6	6	6	2	6	**2**	n	4
Frame Control	Duration/ ID	Address 1	Address 2	Address 3	Sequence control	Address 4	***QoS Control***	Frame Body	FCS or CRC

Fig. 2.4: IEEE 802.11e MAC Frame Format.

2.2.6.1 Enhanced Distributed Channel Access (EDCA)

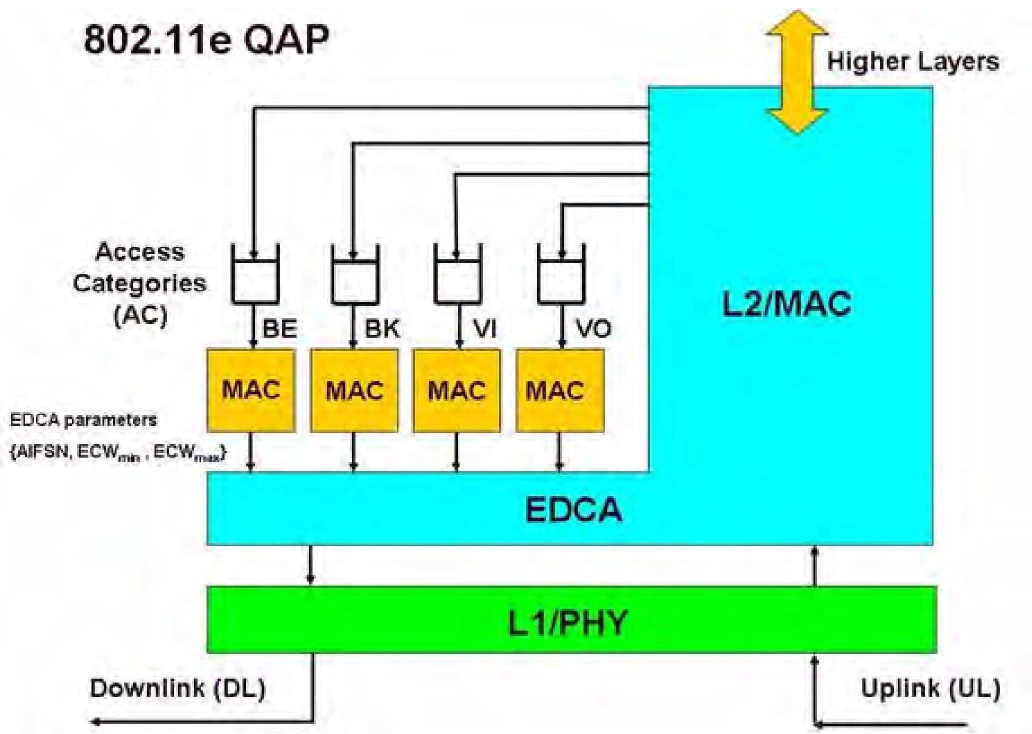

Fig. 2.5(a): IEEE 802.11e Access Categories

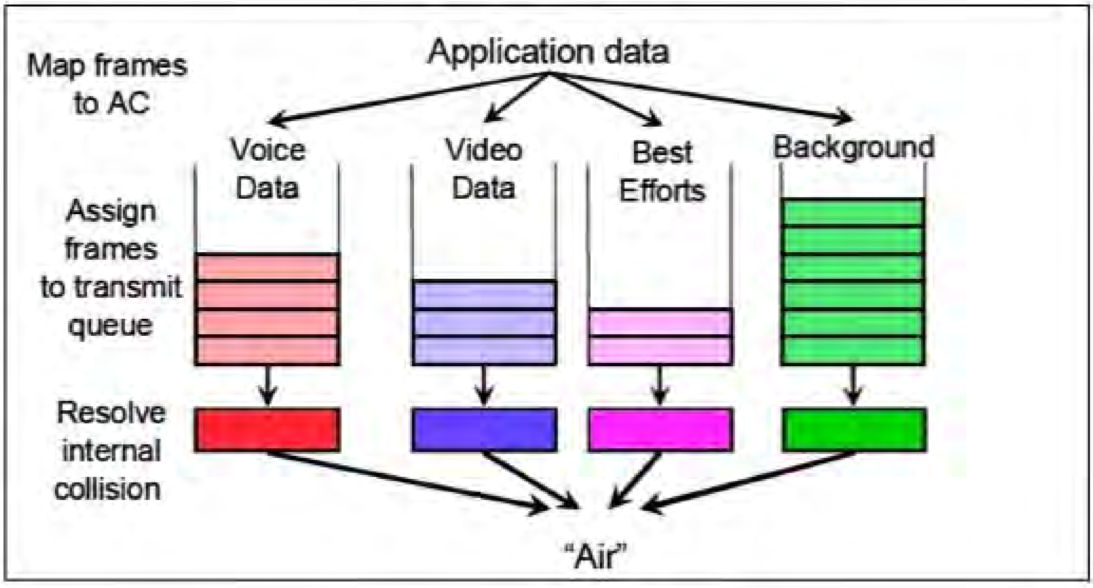

Fig. 2.5(b): Schematic Diagram of Four IEEE 802.11e Access Categories

(Courtesy Aruba Networks)

The IEEE 802.11e standard defines four access categories (ACs) each with its own transmit queue and associated set of AC parameters (Fig. 2.5(a,b,c)). These ACs are labelled voice (*AC_VO*), video (*AC_VI*), best effort (*AC_BE*) and background (*AC_BK*). Applications tag packets to indicate which AC they belong to. These ACs are implemented as four separate queues each with their own CSMA/CA MAC mechanism. Incoming packets are then allocated to one of four independent transmit queues. Two basic priority mechanisms for accessing the channel are used, namely Arbitration Inter-Frame Space Number *(AIFSN),* contention window back-off intervals (ECW_{min} and ECW_{max}). Each AC contends independently for access to the channel based on the above parameters within the QSTA. Stations try to send data after detecting the medium is idle and after waiting a period of time defined by the corresponding traffic category called the Arbitration Interframe Space *(AIFS).* To avoid collisions within a traffic category, each AC independently starts counting down an additional random number of time slots, known as a contention window, before attempting to transmit data. Data frames from the AC with the highest priority have the right to initiate frame exchange sequences onto the wireless medium. Video and voice are the highest priority queues, best effort is medium priority queue, and background is the lowest priority queue.

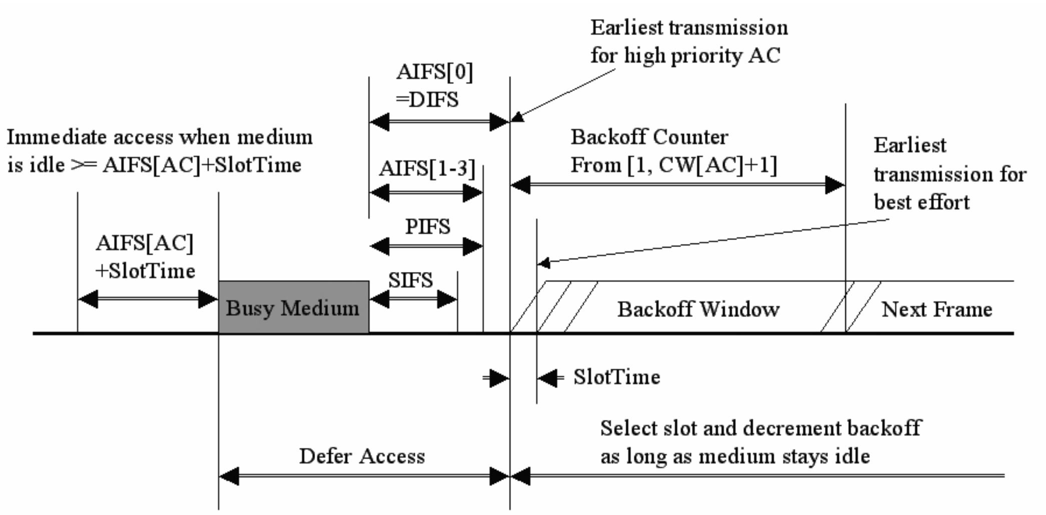

Fig. 2.5(c): IEEE 802.11e EDCA Mechanism

Once the channel is accessed by a QSTA, it is allowed to hold the channel for a certain amount of time which is known as transmission opportunity (*TXOP*). Once a client gains a *TXOP*, it is allowed to transmit for a given time that depends on the AC and the PHY rate. Hence devices operating at higher PHY rates are not penalized when devices that support only lower PHY rates contend for medium access. The *TXOP Limit* can be used to ensure that high-bandwidth traffic gets greater access to the medium.

Broadly speaking, to prioritize a certain traffic type, the values of *AIFSN,* ECW_{min}, and ECW_{max} need to be kept at a lower values compared to other type of traffic [24], i.e.

$$AIFS_{VI,VO} \leq AIFS_{BE,BK} \quad \text{......} \quad (2.4)$$

$$ECW_{\min VI,VO} \leq ECW_{\min BE,BK} \quad \text{......} \quad (2.5)$$

$$ECW_{\max} \leq ECW_{\max BE,BK} \quad \text{......} \quad (2.6)$$

2.2.6.1.1 ***ECW$_{min}$* and *ECW$_{max}$***

A random backoff value is selected in the range 0 to CW. If the first random backoff wait time expires before the data frame is sent, a retry counter is incremented and the random backoff value (window) is doubled (Fig 2.6). This doubling continues until either the data frame is sent or the Maximum Contention Window size is reached. CW is expressed exponentially –

$$CW = 2^{ECW} - 1 \quad \text{......} \quad (2.7)$$

Where, $ECW = ECW_{\min}$ initially.

The minimum contention window is the upper limit (in units of time slots) of a range from which the initial random backoff time is determined. AC with higher priority is assigned a shorter ECW_{min}. The maximum contention window is the upper limit for the doubling of the random backoff value. The doubling continues until either the data frame is sent or the maximum contention window is reached.

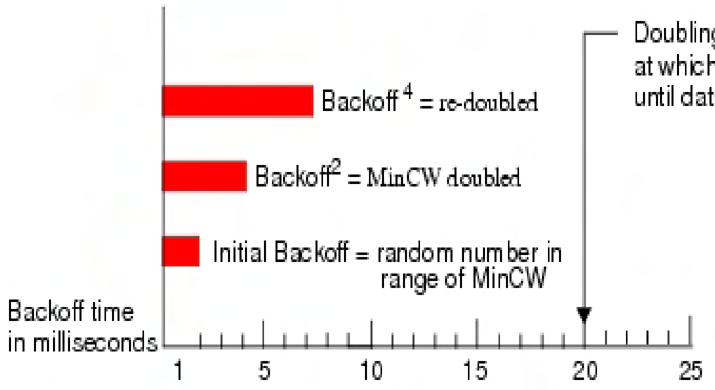

Fig. 2.6: Illustration of Contention Window Doubling

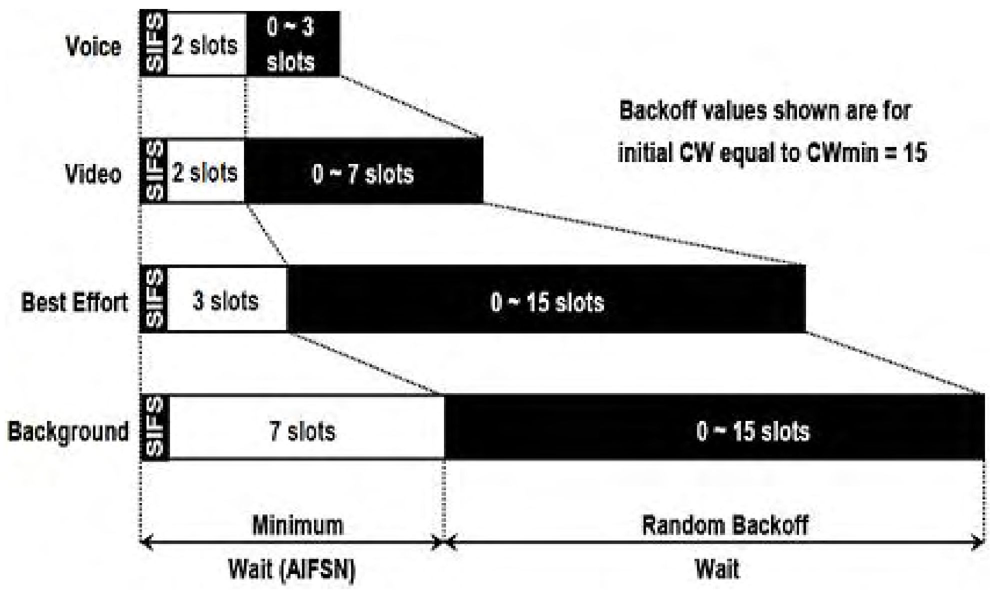

Fig. 2.7: Illustration of *AIFSN* and CW

2.2.6.1.2 Arbitration Inter-Frame Space Number (*AIFSN*)

A higher-priority traffic category will have a shorter *AIFS* than a lower-priority traffic category. Thus stations with lower-priority traffic must wait longer on average than those with high-priority traffic before trying to access the medium.

The duration *AIFS* [25] can be derived from the value *AIFSN* by the relation -

$$AIFS[AC] = AIFSN[AC] * aSlotTime + aSIFSTime \qquad \text{.................} \qquad (2.8)$$

where *aSlotTime* is the slot time, *aSIFSTime* is the *SIFS* time period and *AIFSN* is used to determine the length of the *AIFS*. *AIFSN* specifies the number of time slots (Fig 2.7) in addition to the *SIFS* time period the *AIFS* consists of. The minimum and maximum values for *AIFSN* are 2 and 15 respectively for QSTAs. For QAP the values range from 1 to 15.

2.2.6.1.3 Transmission Opportunity (*TXOP*)

The IEEE 802.11e standard introduced a new MAC mechanism known as the transmission opportunity (*TXOP*) [26]**.** The *TXOP* is specified per AC, is either obtained by an AC in a QSTA by successfully contending for the channel or assigned by the hybrid coordinator (HC). A *TXOP* is defined by a starting time and a maximum duration. During this interval of time, a client station has the right to initiate transmissions on the wireless network without having to re-contend for access and is allowed to transmit multiple MPDUs from the same AC with a *SIFS* time gap between an ACK and the subsequent frame transmission. Therefore *TXOP* provides collision free and contention free transmission period. If a frame is too large to be transmitted in a single *TXOP*, it should be fragmented into smaller frames. The *TXOP* scheme becomes inefficient if there are not many packets present in a winning queue. When there are no more packets to be sent during the *TXOP* interval and the channel becomes idle again, the IEEE 802.11 HC may sense the channel and reclaim the channel after duration of *PIFS* after the *TXOP*.

Devices operating at higher PHY rates are not penalized when devices that support only lower PHY rates contend for medium access. The *TXOP Limit* can be used to ensure that high-bandwidth traffic gets greater access to the medium. System performance can be increased by dimensioning *TXOP Limits* effectively for real time multimedia streams [27,28].

Under the IEEE 802.11e standard, the maximum allowable *TXOP Limit* is 8160 μs with a default value of 3008 μs, in units of 32 μs. The default values for different parameters can be found in Table 2.1. But in the standard it is not optimized for particular traffic types. From the above discussion it comprehensible that by appropriate tuning of these four access parameters (*AIFSN*, ECW_{min}, ECW_{max}, and *TXOP Limit*), it is possible to introduce a relative prioritisation between these queues in winning access to the medium and transmitting their packets. The benefits of QoS become more obvious for high load on the wireless LAN, keeping the loss, delay, and jitter for multimedia traffic types within an acceptable range. Prioritization is an important mechanism to provide QoS. By providing QoS to wireless multimedia streams bandwidth utilization can be made more effective.

Table 2.1: Default IEEE 802.11e Parameters According to The Standard

AC	CW_{min}	CW_{max}	*AIFSN*	*TXOP* Limit (802.11b)	*TXOP* Limit (802.11a/g)
AC_BK	CW_{min}	CW_{max}	7	0	0
AC_BE	CW_{min}	CW_{max}	3	0	0
AC_VI	$(1+CW_{min})/2-1$	CW_{min}	2	6.016 ms	3.008 ms
AC_VO	$(1+CW_{min})/4-1$	$(1+CW_{min})/2-1$	2	3.264 ms	1.504 ms

2.2.6.2 IEEE 802.11e HCCA

The hybrid coordination function (HCF) controlled channel access (HCCA) is the enhanced form of the PCF and operates on top of the EDCA.

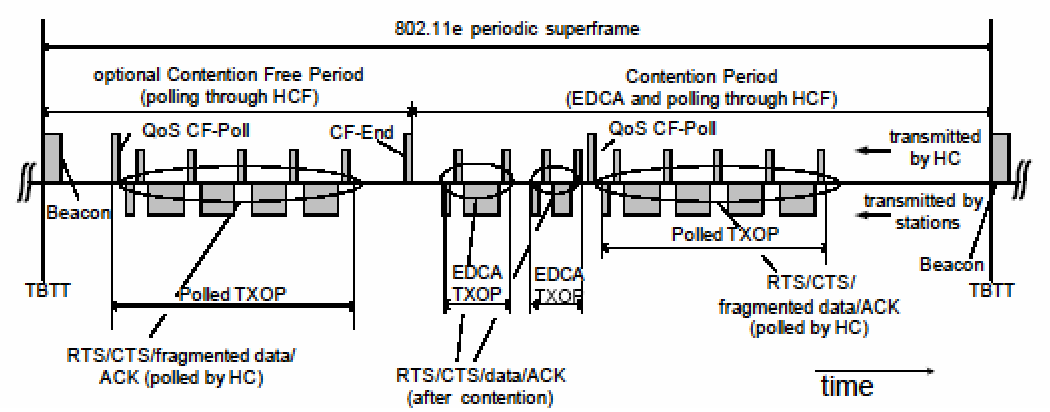

Fig. 2.8: HCCA Operation

The HC is co-located at the QAP. A QSTA requests the HC for reservation of *TXOP*s based on its QoS requirements, both for its own transmissions as well as for transmissions from the QAP to itself. If the HC accepts the request (based on its admission control policy), *TXOP*s for both the QAP and the QSTA are scheduled. As shown in Fig. 2.8, for transmissions from the QSTA, the HC polls the QSTA based on the parameters supplied by the QSTA at the time of its request. The QAP directly obtains *TXOP*s from the HC for transmissions to the QSTAs and delivers the frames to the QSTA, again based on the parameters supplied by the QSTA. As everything is predetermined upon registration, HCCA is able to guarantee bandwidth, jitter and latency, which is otherwise a difficult challenge in a mixed data and multimedia environment. The operation of HCCA is not considered in this thesis. Instead the thesis will focus on the EDCA mechanism of the IEEE 802.11e standard only.

2.2.6.3 IEEE 802.11n

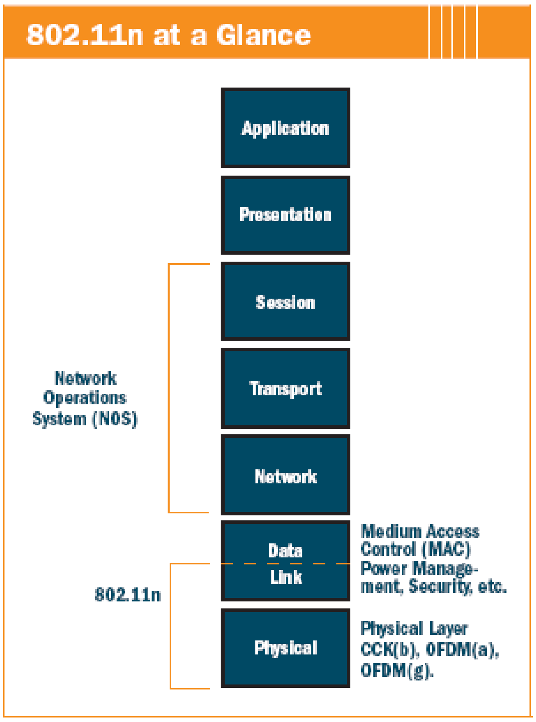

Fig. 2.9: IEEE 802.11n Requires a New Physical Layer, Along with Changes to the Bottom Half of the Data Link Layer (i.e. the MAC)— the Other Aspects of the Wi-Fi Network Are Untouched

This standard was ratified in September, 2009 and aims to achieve much higher data rates than previous IEEE 802.11 standards by modifying both the PHY layer and MAC sub-layer (Fig. 2.9). It was designed to support transmission speeds up to 150 Mbps per stream and up to four streams (i.e. 600 Mbps over 40 MHz bandwidth). It includes the QoS mechanisms introduced by IEEE 802.11e and is backward compatible with IEEE 802.11a/b/g networks.

The PHY layer improvements include using Orthogonal Frequency Division Multiplexing (OFDM) modulation coupled with Multiple Input Multiple Output (MIMO) technology (Fig. 2.10) to increase data rate in both the 2.4 GHz and 5

GHz bands [20,29]. IEEE 802.11n radios define data rates based on numerous factors including modulation, the number of spatial streams, channel size, and the guard interval. A combination of these multiple factors is known as a *Modulation and Coding Scheme* (MCS).

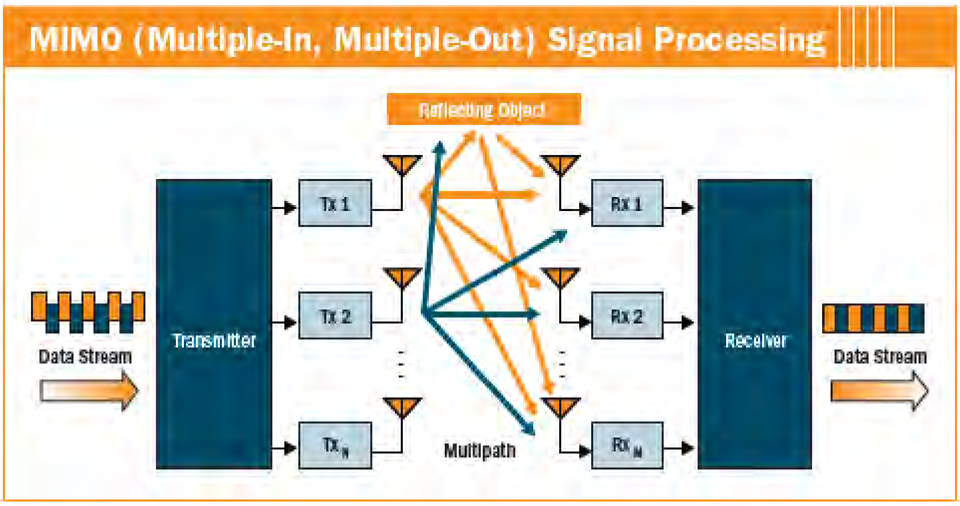

Fig. 2.10: Utilizing MIMO, 802.11n Can More Than Double Existing Data Rates, Depending upon the Number of Antennas Being Used.

OFDM is a FDM technique for transmitting large amounts of digital data over a radio wave by transmitting the binary data over multiple sub-carriers to the receiver.. IEEE 802.11n uses multiple transmit and receive antennas to transmit the same data stream to improve signal reception and is expected to use non-overlapping channels with channel bandwidths of 20 and 40 MHz (Fig. 2.11 and 2.12). Channel bonding (Fig. 2.13) is used in this standard where two adjacent contiguous 20 MHz channels are combined into a wider 40MHz channel. In the 2.4 GHz band there are only 3 non-overlapping 20 MHz-channels. Within the 5 GHz band there are 24 non-overlapping channels and consequently a higher degree of freedom. The 5 GHz range contains less interference than the 2.4

GHz. With less interference, any device operating in 5GHz will have a much cleaner signal than one operating in the congested 2.4GHz range.

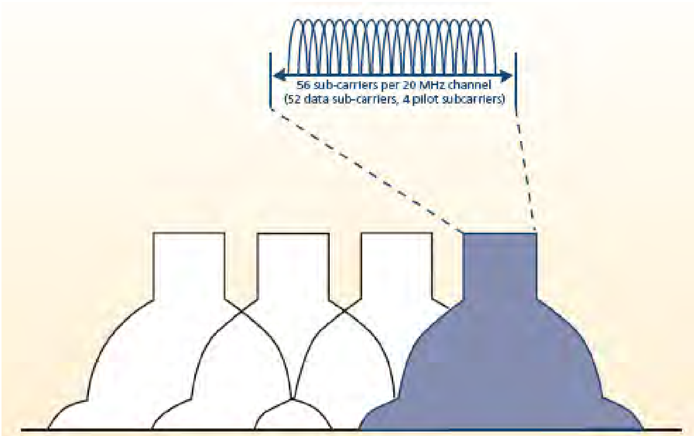

Fig. 2.11: 20 MHz OFDM Channel

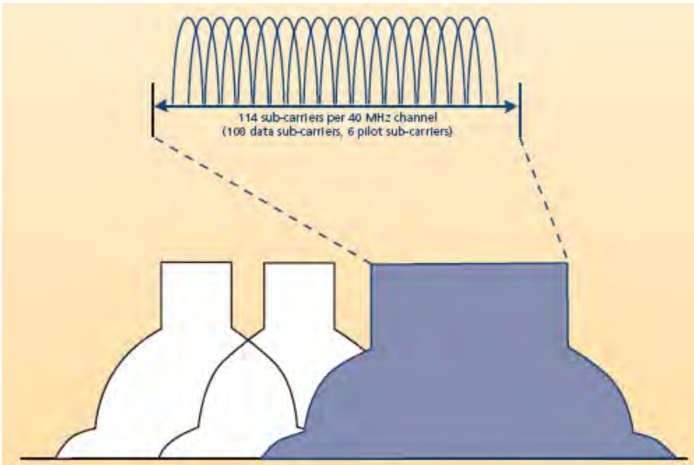

Fig. 2.12: 40 MHz OFDM Channel

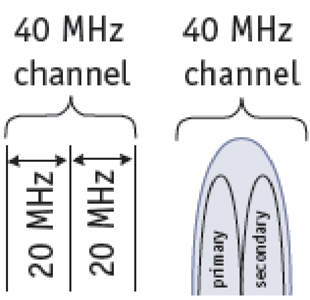

Fig. 2.13: Channel Bonding

The WLAN standard provides up to four spatial data streams and thus up to a fourfold bit rate. Doubling the number of spatial streams from one to two effectively doubles the raw data rate. A guard interval is a set amount of time between transmissions, designed to ensure that distinct transmissions do not interfere with one another. A guard interval of 800 ns was set in the original IEEE 802.11 specifications which was longer than was needed in many environments. A shorter guard interval (400 ns) was added as an option in the 802.11n specification to allow for higher data rates where a long guard interval is not required.

In order to increase the MAC layer throughput efficiency in WLANs, frame aggregation and block acknowledgement schemes have been added in this standard. In the current IEEE 802.11 MAC layer, an STA wait has to gain access to the medium for each and every packet it has to send. When the frames are small, the waiting time results in severe underutilization of the wireless medium. Frame aggregation is a mechanism used to combine multiple frames into a single frame transmission allowing an increase in overall performance. Multiple frame payloads (MSDUs) can be aggregated into a single frame known as an *Aggregate MAC Service Data Unit (A-MSDU)*. As pictured in Fig. 2.14, multiple 802.11 frames (MPDUs) can be aggregated into a single frame known as *Aggregate MAC Protocol Data Unit (A-MPDU)*. Also, to maximize the throughput efficiency of this method, the maximum frame size is increased, allowing longer frames.

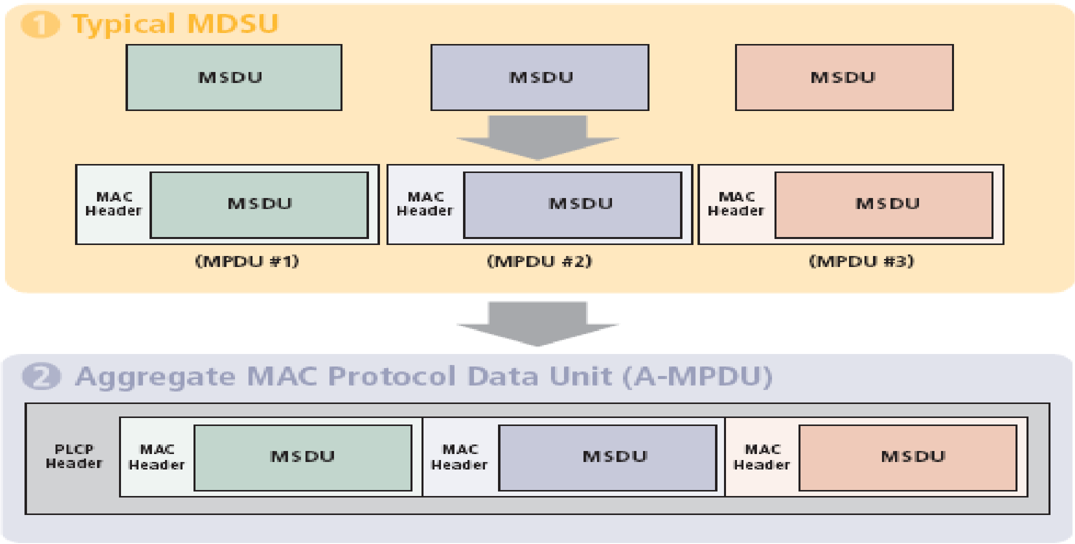

Fig. 2.14: Frame Aggregation

The IEEE 802.11 standard requires that all 802.11 unicast frames be followed by an ACK frame. Block acknowledgments are needed to verify the delivery of the multiple MPDUs that are aggregated inside a single A-MPDU transmission. When using the A-MPDU frame aggregation method each A-MPDU contains multiple frames, and each of the individual MPDUs must be acknowledged. This is accomplished by using a *multiple traffic ID block acknowledgment (MTBA)* frame. The use of block acknowledgements decreases MAC layer overhead and thus increases throughput and reliability.

Another improvement to the IEEE 802.11n MAC is Reduced Interframe Spacing (RIFS). This is a change from the current standard, where Short Interframe Spacing (SIFS) is used. RIFS greatly minimizes the space between packets that are being sent out over the air, thereby decreasing unusable dead time.

Table 2.2: Comparison of Different IEEE 802.11 Standards

	802.11b	802.11a	802.11g	802.11n
Standard Approved	1999	1999	2003	2009
Maximum Data Rate	11 Mbps	54 Mbps	54 Mbps	600 Mbps
Frequency Band of Operation	2.4 GHz	5 GHz	2.4 GHz	2.4/ 5 GHz
Non-Overlapping Channels	3	24	3	3/24
Number of Spatial Streams	1	1	1	1,2,3, or 4
Modulation Type	DSSS, CCK	OFDM	DSSS, CCK, OFDM	DSSS, or CCK, OFDM
Channel Width	20 MHz	20 MHz	20 MHz	20/40 MHz
Available Bandwidth	83.5 MHz	580 MHz	83.5 MHz	83.5/580 MHz

2.2.7 Different Types of Losses

There are many WLAN performance metrics described in the literature, e.g. throughput, delay, loss rate, jitter etc. A novel QoS aware MPEG-4 video delivery algorithm for IEEE 802.11b will be proposed in chapter 4 and validated in chapter 5 which will primarily be dealing with the performance aspect related to frame losses. There are three ways in which video frames can get lost on a WLAN. These are – MAC collisions arising from contention for access, buffer overflow due to an insufficient availability of transmission opportunities to satisfy the incoming video frames and transmission errors due to noise and interference present in the medium. These are discussed in detail in the following sections.

2.2.7.1 MAC Collision Loss

IEEE 802.11b WLANs provide a best effort data service where the wireless medium is shared. As the MAC is based upon random access where each station initializes its BC by randomly selecting an integer from a finite contention window, there is a finite probability that two or more stations may select the same initial BC value resulting in simultaneous transmission when their respective BCs reach zero giving rise to a collision. Therefore, as the number of stations contending for access increases the probability of collision also increases and even under ideal channel conditions (i.e. a noiseless channel) there will be packet loss due to MAC collisions.

2.2.7.2 Buffer Overflow Loss

Before a station starts transmitting, its MAC buffer holds a video frame while it is waiting for a transmission opportunity. As long as a station wins sufficient transmission opportunities its MAC buffer remains empty and the buffer never fills to exceed its capacity and hence packets are never lost. If a station does not win enough transmission opportunities then as the video frames arrive at the buffer they are enqueued. A station can be described as being in saturation when it always has a frame to transmit in its buffer, i.e. the station is always contending for access. Hence when many stations are contending to access the wireless medium, by measuring the station buffer occupancies, an indication of the onset of saturation can be obtained.

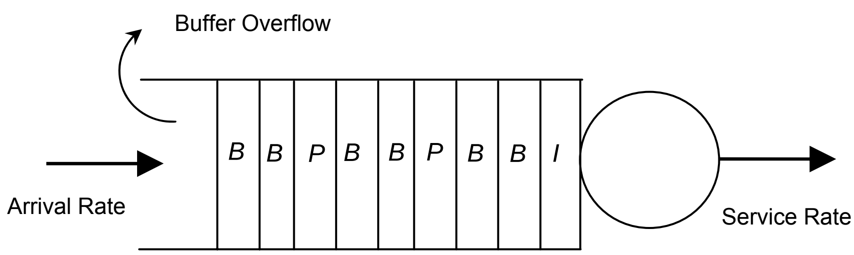

Fig. 2.15: A Typical FIFO Buffer Queue.

As discussed in section 2.2.4, to effectively study the dynamics of the FIFO buffer occupancy it is necessary to analyse the behaviour of two particular aspects of the queuing mechanism, namely the arrival rate (R_A) and the service rate (R_s) as shown in Fig. 2.15. If the average rate of video frame arrival is much greater than the service rate, the AP buffer will fill up quickly and overflow leading to the loss of video frames. To avoid buffer overflow and the resulting drop in video QoS, it is required that on average the service rate be greater than the arrival rate. In the short term, if the arrival rate exceeds the service rate the buffer should be capable of temporarily storing video frame packets until they can be serviced. But the average service rate has to be greater than or at least equal to the average arrival rate to avoid buffer overflow in the long run.

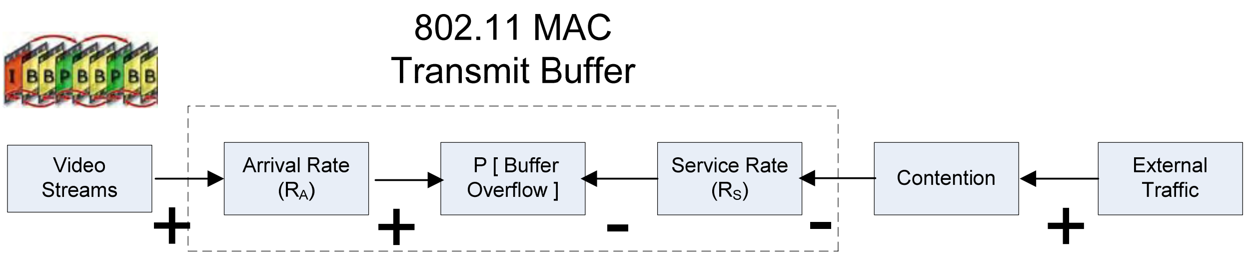

Fig. 2.16: Modelling Buffer Occupancy

For the case of a single server queue with stationary arrival and service rates, a stable system would require that –

$$E[R_A] \leq E[R_S] \qquad (2.9)$$

Where $E[\ ]$ is the expectation operator.

Although the arrival rate can be considered to be a constant, the service rate is essentially a random process due to the CSMA/CA mechanism used in the MAC. Therefore there is a finite probability of buffer overflow due to the buffer having a finite capacity. Therefore, even in a stable system the possibility of buffer overflow always exists. The Buffer occupancy has been modelled in Fig 2.16.

2.2.7.3 Transmission Loss

In WLANs multipath propagation, noise, and interference can also cause frames to become corrupted at the receiver. This type of frame loss is known as transmission loss. On real networks packet loss due to transmission errors is inevitable and frame re-transmission mechanism is used to address transmission loss. Retransmissions require feedback from the receivers, specifying frames required for transmitting again. IEEE 802.11 uses a retransmission mechanism to improve the reliability of unicast traffic, but provides no reliability support to broadcast and multicast traffic. In IEEE 802.11 unicast, a station transmits the packet and waits for an ACK. If the sender does not receive an ACK (e.g., due to poor channel condition), it retransmits the packet using binary exponential back-off, where its contention window is doubled every time after a failed transmission until it reaches its maximum value, denoted as CW_{max}. The retransmission limit thus affects the packet loss due to transmission errors in the medium. Also line rate adaptation in IEEE 802.11 is based upon transmission loss. However, this mechanism [30,31,32] has not been considered in this thesis.

2.3 Video

There are various video display standards for analog and digital systems. Analog video standards were the original standards and the three main standards are NTSC, PAL, and SECAM. The NTSC standard is primarily used in North America, parts of South America and Japan, PAL is used in Europe, Asia, Australia, etc. and SECAM is used in France, Russia and parts of Africa.

There are currently three dominant digital standards available: ATSC, DVB, and ISDB. ATSC is used in North America, DVB in Europe, and ISDB in Japan. ATSC has replaced the analog NTSC television system in USA on February 17, 2009. The European Union and Canada have set a union-wide target date of 2012 and August 31, 2011 for digital switchover [33]. Switch-off has already been completed in five EU member states (Germany, Finland, Luxembourg, Sweden and the Netherlands) [34]. Australia's switch-off is planned for 2013 and India and Russia for 2015.

The important characteristics of digital video streaming are the frame rate, bit rate, aspect ratio, and resolution.

Frame rate (fps): This specifies the number of frames (i.e. images) per second present in a video stream. Frame rate is the way we perceive motion. 25 fps is considered sufficient to capture smooth motion. The PAL and SECAM standards specify 25 fps, while NTSC specifies 29.97 fps. The ISDB standard supports 30 fps. The ATSC and DVB systems support a number of different frame rates with the maximum being 60 fps.

Bit rate (bps): This is the measure of the information content per unit time in a video stream. A higher bit rate can be interpreted as having a better video quality. Variable bit rate (VBR) and constant bit rate (CBR) strategies are often used depending on the type of application, network topology etc.

Aspect ratio: This is the ratio of the width to the height of the video screen. For HDTV and traditional television it is 16:9 and 4:3 respectively.

Resolution: This is the size of an image which is measured in pixels for digital video, or horizontal and vertical lines of resolution for analog video. The pixel (from "picture element") is the basic unit of programmable colour on a computer display or in an image.

Generally digital video is compressed by reducing the spatial and temporal redundancy to increase the efficiency. The most popular encoding standards are MPEG-1, MPEG-2, MPEG-4, H.264 etc [35].

MPEG-1

MPEG-1 is an early standard for the compression of audio and video. The MP3 audio format is a well known part of the MPEG-1 standard. Part 1 of the MPEG-1 standard covers Systems, part 2 covers Video and part 3 covers Audio. The Systems part specifies how to maintain synchronization between the different contents and the logical layout and methods used to store the encoded audio, video, and other data into a standard bit-stream. The MPEG-1 standard defines the bit-stream, and decoder function, but does not define how MPEG-1 encoding is to be performed.

MPEG-2

MPEG-2 [36] standard is backward compatible with MPEG-1. MPEG-2’s part 1 section defines two container formats - transport stream and program stream, part 2 describes video, and part 3 is concerned with audio. The transport stream is designed to carry digital video and audio over lossy media while the program stream is designed for somewhat more reliable media such as DVDs, SVCDs, optical disks etc. The video section is similar to the previous MPEG-1 standard plus it also provides support for interlaced video. The audio section enhances MPEG-1's audio by allowing the coding of audio data with more than two channels.

MPEG-4

MPEG-4 dramatically advances audio and video compression, enabling the distribution of content and services from low bandwidths to high-definition quality across broadcast, broadband, wireless and packaged media. MPEG-4 consists of closely interrelated but distinct individual Parts (Fig. 2.17), that can be individually implemented (e.g., MPEG-4 Audio can stand alone) or combined with other parts. The basis is formed by Systems (part 1), Visual (part 2) and Audio (part 3). DMIF (Delivery Multimedia Integration Framework, part 6) defines an interface between application and network/storage. Conformance (part 4) defines how to test an MPEG-4 implementation, and part 5 gives a significant body of Reference Software, that can be used to start implementing the standard, and that serves as an example of how to do things. Part 7 of MPEG-4 defines an optimized video encoder (in addition to the Reference Software, which is a correct, but not necessarily optimal implementation of the standard)

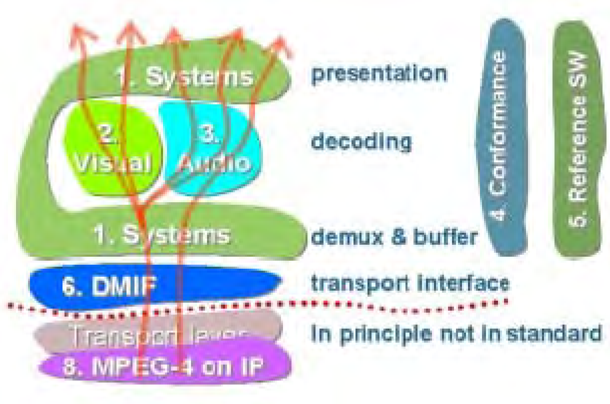

Fig. 2.17: The Parts of MPEG-4. The Arrows Represent the Flow of Bits through the MPEG-4 System.

Parts of the MPEG-4 Standard [37]

Part 1. Systems

This part of the standard deals with scene description and identification of its constituent objects, synchronisation and multiplexing of Elementary streams (ESs), buffer management, decoder models and IPMP.

Part 2. Visual

Techniques used for natural and synthetic video and image coding such as compression, error resilience, facial and body animation and 2D and 3D meshes are defined in this section of MPEG-4.

Part 3. Audio

Coding of natural and synthetic audio objects is specified in part 3.

Part 4. Conformance Testing

Defines the conformance conditions to ensure compatibility of ESs and devices and is used to test MPEG-4 implementations.

Part 5. Reference Software

Software corresponding to various other parts such as Part 2 and Part 3, of the standard is included as part of the standard to allow study and implementation of MPEG-4 compliant products. ISO waives the copyright of the code.

Part 6. Delivery Multimedia Integration Framework (DMIF)

This part defines a protocol for the management of multimedia streaming over a generic transport layer.

Part 7. Optimised Reference Software

Part 7 provides a reference implementation of the visual tools that includes some features not considered in Part 5, such as: fast motion estimation, fast global motion estimation and fast and robust sprite generation.

Part 8. Carriage of MPEG-4 Contents over IP Networks

Specifications of a framework for the carriage of MPEG-4 over IP networks are presented in this part. It defines guidelines for the design of relevant *Request For Comments* (RFC) Documents. Related SDP rules and MIME types are also discussed in this section.

Part 9. Reference Hardware Description

In Part 9 descriptions of the main MPEG-4 coding tools in *Hardware Description Language* (HDL) form are presented.

Part 10. Advanced Video Coding (AVC)

Part 10 explains the video syntax and coding tools for a joint project between MPEG and the International Telecommunication Union - Telecommunication standardization sector (ITU-T). AVC is intended for a broad range of natural video applications such as broadcast video.

Part 12. ISO Base Media File Format

In this part, guidelines for a file format used to contain time-based media are specified. The file format is designed to be flexible and support both local and network access.

Part 14. MP4 File Format

The MP4 file format defines the storage of MPEG-4 content in files. It is a more versatile format than the ISO Base Media File Format, permitting a wide variety of usages, such as editing, display, interchange and streaming.

Part 15. AVC File Format

Part 15 of the standard defines how to store content from Part 10 of the standard in the File Format prescribed in Part 14.

Part 16. Animation Framework eXtension (AFX)

This section proposes a general organization of synthetic models in terms of geometry, modeling, physical, biomechanical, behavioural and cognitive components for interactive multimedia content including computer games and animation.

Part 18. Font Compression and Streaming

Specifications for font data representation, compression, streaming and providing an efficient mechanism to embed font data in MPEG-4 encoded presentations are presented in Part 18.

Part 19. Synthesized Texture Stream

Part 19 deals with the transmission of synthesised texture data and it defines the coded representation for synthesized texture data streams and also the associated data structures and animation methods.

From the above discussion it can be stated that many of the features of MPEG-1 and MPEG-2 are included in MPEG-4 standard. Some new features have been added, for example coding efficiency has been improved, media data (voice, video, and audio) encoding is possible, error resilience techniques are included, and MPEG-4 has the ability to interact with the audio-visual scene generated at the receiver. In MPEG-4 format, the data volume is only about 1/11th the size of the original MPEG-2 video for similar quality i.e. MPEG-4 standard provides high video quality at low data rates.

A MPEG-4 file consists of 2 constituent parts: a header and a payload. The header contains data relating to the frame rate, width, height and frequency of the video file; and the payload contains the video frame data. This standard satisfies the needs of the industry players (e.g. developers, service providers, end users) by allowing greater reusability and flexibility. MPEG-4 provides a generic QoS descriptor for different MPEG-4 media. MPEG-4 also provides specifications to transport content over IP paving the way for multimedia streaming through wired and wireless networks.

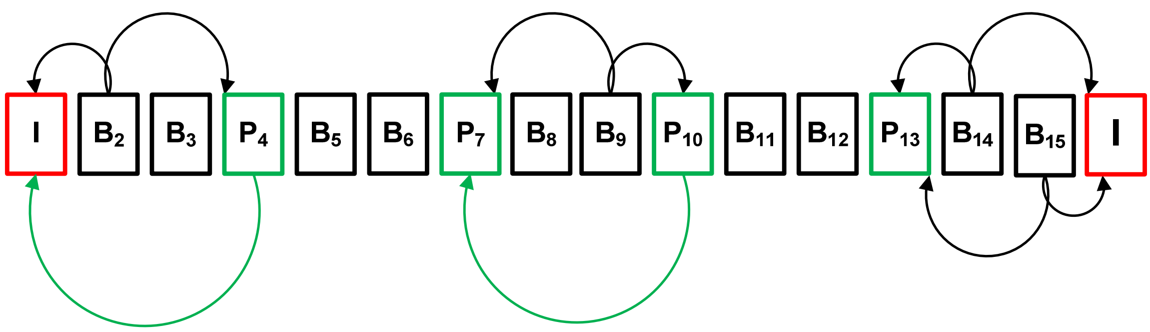

Fig. 2.18: Typical MPEG Encoding Pattern

Under the MPEG standards images can be encoded into three frame types: intra-frames (*I* frames), forward predicted frames (*P* frames), and bi-directional predicted frames (*B* frames) as shown in Fig. 2.18. *I* frames are encoded as self-contained JPEG images [38,39,40], i.e. they are independent of any past or future image in the video sequence. They usually occur once or twice per second of video stream. *P* frames are encoded relative to the past reference *I* or *P* frame. So they contain information relating to what has changed since the previous frame by calculating the block-by-block difference. The information contained in *B*-frames is based upon the previous and succeeding *I* or *P* frames in the video stream. There is another type of frame called a *D* frame which is used for fast-forward and rewind.

A Group of Pictures (GOP) size is defined as the length between two successive *I* frames. A GOP size of 15 means that in a video sequence there is one *I* frame for every 14 non- *I* frames, i.e. for a combination of *P* and *B* frames. Encoders choose GOP sizes dynamically. *I* frames are the most important for reconstructing video as they contain the most visual data. They are particularly important because they prevent the propagation of errors from previously damaged or incorrectly predicted frames into subsequent frames. An MPEG-4 file constructed of *I* frames alone would have excellent video quality but would also have poor compression. The loss of a *B* frame does not have as great an effect on the final video quality compared to the loss of an *I* frame during transmission.

H.264 is also known as MPEG-4 Part 10, or MPEG-4 AVC. This standard provides good video quality at substantially lower bit rates and is applied to a wide range of applications on a wide variety of networks and systems, including multimedia telephony systems, variable resolution video, broadcast, DVD storage, RTP/IP packet networks etc.

Table 2.3 gives a comparison of MPEG-4 against most commonly used multimedia formats on the Internet today [41].

Table 2.3: Comparison of MPEG-4 against Most Commonly Used Multimedia Formats on the Internet

	MPEG 4	**Windows Media**	**Real**	**Flash**
Audio/Video Codec	Standards based; multivendor support.	Proprietary	Proprietary, but supports automatic download of MPEG-4 plug-in.	Proprietary + proprietary Real and QuickTime formats.
Interactivity	Highly interactive.	Limited	Yes, via SMIL.	Highly interactive.
Digital Rights Management	Interfaces to proprietary DRM. More interoperable DRM under development in MPEG-4 and MPEG-21	Microsoft DRM	Content access control	No
Real-time stream control	Yes	Yes	Yes	No
Synchronization	Audio, video and all other objects can be tightly synchronized with high accuracy	Tight synchronization between audio and video	Tight synchronization between audio and video	No synchronization between scene and streams
Broadcast capable	Yes, including interactive features	A/V only	Scene must be unicast	No
Object model support	Video/audio and rich 2D/3D mixed media, synthetic graphics. DRM on separate streams.	Audio/Video only	Video/audio and mixed media through SMIL based protocol. No streaming of mixed media.	Video/audio and mixed media through proprietary protocol.
Graphic Objects	Yes	No	No	Yes
Transport	Support exists for HTTP, UDP, RTP/RTSP, MPEG-2TS, mobile	HTTP, UDP, RTP/RTSP, mobile	HTTP, RTP/RTSP, mobile	HTTP

2.4 Video Streaming

Streaming is the process of playing out a file while it is still downloading. It allows a user to experience multimedia content - video, voice, animation etc. - as it is being downloaded. Multimedia content has significantly different characteristics compared to data traffic. Its packet sizes tends to be much larger and it requires greater network bandwidths for its transmission. Moreover it tends to exhibit a large variation in throughput. According to Bernstein Research, downloading half an hour of television-quality video on the web consumes more bandwidth than sending and receiving 200 emails per day over an entire year [42]. Also various purpose built protocols are required for streaming media over network. Content can be either analog or digital. Streaming media may be real-time or on-demand.

On demand streams are stored on a server and upon user request the content is transmitted. The user may then play the stream locally (with a suitable player) or download it for viewing. Real-time streams are only available at one particular time, i.e. when the event is occurring in real time. The user may record it in real time provided that the necessary hardware and software are present.

Currently MPEG-4, Windows Media, and H.264 are the most popular formats for streaming video over wired and wireless networks. Another free and open compression standard called *Theora* [43] has been standardized by the Xiph.org Foundation, but is still in development.

The video channels for communication may also be static or dynamic, packet-switched or circuit switched, may support a constant or variable bit rate transmission, and may support some form of QoS, or may only provide best effort support. The specific properties of a video communication application strongly influence the design of a system that is used for video streaming.

The popularity of streaming wireless multimedia is continuing to grow. It is predicted to be one of the highest revenue generating technologies in the near future [44]. With such a great economical potential, product development and standardization processes have been accelerated by both academic and commercial bodies. For real time multimedia applications tolerable delay rate is in the order of 150 ms - 400 ms. Table 2.4 shows the typical QoS requirements for multimedia services as defined by ITU-T [45,46].

Table 2.4: QoS Requirements for Multimedia Services

Class	Application	One way transmission delay	Delay variation	Packet loss rate
Real time	VOIP, Video conferencing	<150ms(preferred) < 400ms (limit)	1 ms*	1% (video) 3%(audio)
Streaming	Streaming audio and video	Up to 10s	1ms*	1%
Best effort	Email, file transfer, web browsing	Minutes to hours	N/A	Zero
*Playout buffer (jitter buffer) can be used to compensate for delay variation				

Video streaming has different requirements for delay, jitter (variation in delay), and loss characteristics compared to other data services. Multimedia applications typically impose some packet loss and delay requirements. When several applications try to access the same bandwidth, the ones that are intolerant to time delays and bandwidth fluctuations may not function properly. Factors which affect the quality of video streaming over WLANs are described in detail in section 2.6.

2.4.1 Video Streaming Solutions

There are several commercial and open source platforms and solutions available for streaming multimedia over networks. They are described below-

2.4.1.1 Commercial Video Streaming Solutions

- Apple Quicktime Streaming Server.

  QuickTime Streaming Server exploits the RTP/RTSP open standard to deliver live or prerecorded content in real time over the Internet. It ships with Mac OS X Server. Apple’s video streaming solution is also capable of HTTP live streaming.

- Microsoft Windows Media Server.

  A Windows Media server is designed specifically for streaming on-demand and live digital media to clients. It provides high-quality streaming over a wide range of bandwidths to Windows Media Player and to Web browsers that use the Windows Media Player 9 Series ActiveX control or the Microsoft Silverlight browser plug-in. It is especially useful for streaming large amounts of data over busy, congested networks and low-bandwidth connections. Streaming uses bandwidth more efficiently than downloading because it sends data over the network only at the speed that is necessary for the client to render it correctly. This helps prevent the network from becoming overloaded and helps maintain system reliability.

- RealNetworks Helix Server.

This solution streams multi-format, including Flash, H.264, 3GPP, MP4 and delivers to multi-screens, including iPhone, iPad, Android, and PCs. It operates on Windows, Linux or Solaris 64-bit platforms

- Intel IP Services: P Multimedia Subsystem, IPTV, Mobile TV
- Adobe (Macromedia) Flash Video Server

Adobe Flash Media Server 4.5 software now delivers media to multiple platforms — including Apple iOS devices — with a choice of powerful protocols that can save significant bandwidth costs and lighten network load.

There are some other companies which also provide video streaming solutions -

- Accordent Solutions
- Destiny Media Technologies Clipstream
- Forbidden Technologies FORScene
- TurnStyle Andromeda (mp3 streaming)
- VideoVista
- VX30 Streaming Video Solutions

2.4.1.2 Free and Open Source Video Streaming Solutions

In one case Darwin Streaming Server (DSS) [47] and VideoLAN Client (VLC) [48] were used as a streaming server and video client respectively. There are are several open source streaming servers available – e.g. Helix from Real [49], Darwin Streaming Server from Apple etc. DSS is an open-source, standard-based streaming server that is compliant with MPEG-4 standard profiles, ISMA streaming standards and all IETF protocols. The DSS streaming server system is a client-server architecture where both client and server consist of the RTP/UDP/IP stack with RTCP/UDP/IP to relay feedback messages between the client and server. The client can be any player that is capable of playing out MPEG-4 content. Other open source video streaming solutions include-

- VideoLAN Server (VLC).
- MPEG4IP
- Darwin Streaming Media Server: Open Source version of Apple´s Quicktime Streaming Server
- FFServer (parte de FFMPEG)
- AMpache - Web-based audio file manager
- ePresence
- FreeCast (peer to peer streaming)
- GNUMP3d Streaming MP3 / Media Server
- IceCast
- Logitech SlimDevices SlimServer
- SHOUTcast

- UnReal Server
- Zina (mp3 streamer, open source equivalent to Andromeda)

2.5 Quality of Service (QoS)

According to Cisco, QoS maybe defined as "*the capability of a network to provide better service to selected network traffic over various technologies*" in order to "*provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic such as video, voice and gaming etc.), and improved loss characteristics*" [50]. The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency and improved loss characteristics as required by real-time and interactive traffic. Ensuring a QoS will involve the following functions:

- Resource allocation: mechanism to allocate bandwidth and buffer space to a new traffic flow, so that all traffic flows can get their QoS.
- Congestion Control: mechanisms to react to increases in usage of buffer spaces and bandwidth.

QoS for multimedia streaming applications can be subdivided into two key parts: the Quality of Delivery (QoD) and the Quality of Experience (QoE).

The QoD relates to the end-to-end delivery of the multimedia stream over the network in terms of delay, packet loss, throughput, and jitter (delay variation) etc. [51]

The ITU-T defines QoE as a "*measure of the overall accept-ability of an application or service, as perceived subjectively by the end-user*" [52]. It can also be defined in terms of the measured distortion of the multimedia stream from a user’s perspective [53]. According to ITU-T Rec. P.862, QoE can be measured directly through user tests and is expressed in terms of the Mean-Opinion-Score (MOS) on a five-point scale ranging from 1.0 (bad quality) to 5.0 (excellent quality). Other methods to determine QoE include quality degradation models (like the E-model, ITU-T Rec. G.107), instrumental metrics (e.g. PESQ, ITU-T Rec. P.862), or neural network approaches, for instance Rubino’s PSQA method [54]. QoE is complex owing to a large number of independent factors including the encoding configuration, packetisation schemes, error concealment and correction techniques, as well as the nature of the transmission impairments. Moreover, there is a significant degree of uncertainty regarding the impact of a transmission impairment event introduced as a consequence of the spatio-temporal nature of video streaming.

A late 2010 WLAN report [55] conducted by Webtorials identified reliability, security and performance being the three most important WLAN characteristics from the users perspective (Fig. 2.19). QoS is an important issue in wireless networks. IEEE 802.11 WLANs use a shared medium. Different types of traffic are transmitted over this network. Voice, video, and data are converged in packet-based networks including WLANs. Considerable effort is required to ensure the acceptable performance of a network where variable length packets (whose characteristics are also different) are being transmitted. This convergence provides opportunities to enhance network agility and productivity, eliminate excess infrastructure, and reduce costs. Hence the need to address the QoS issue becomes extremely important.

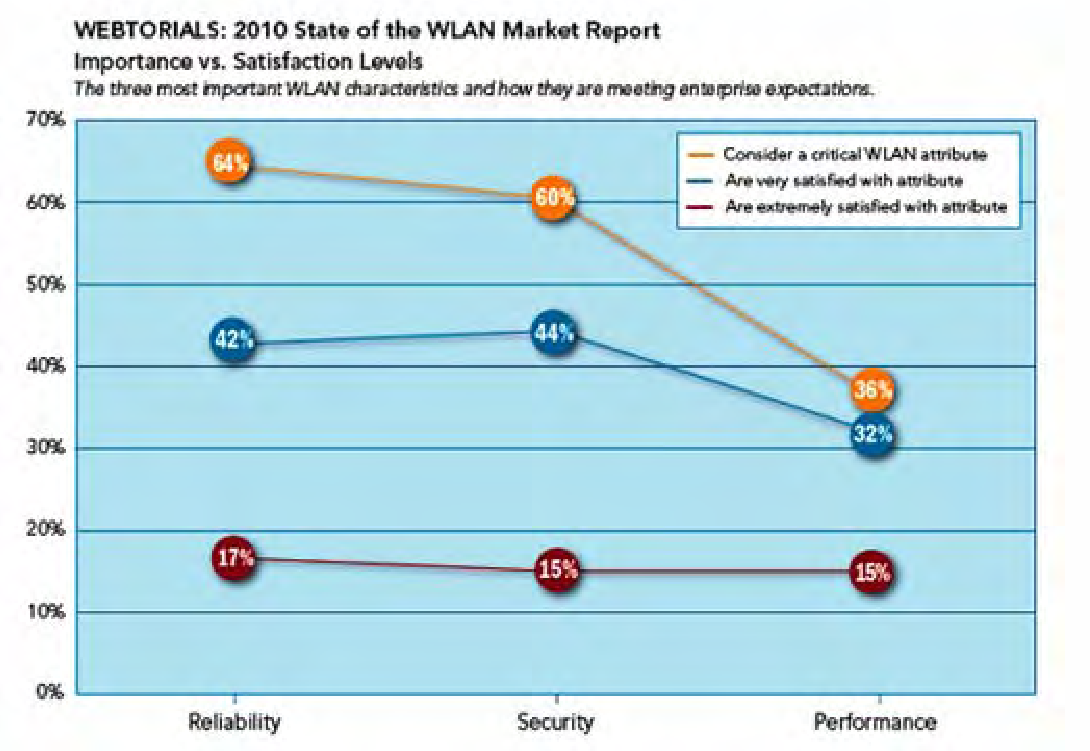

Fig. 2.19: The Three Most Important WLAN Characteristics (Webtorials)

QoS is the result of a set of techniques employed to ensure proper end-to-end network treatment of various traffic types. It refers to providing consistent, predictable data delivery service, i.e. satisfying the customer application requirements. Providing QoS means providing real-time (e.g., video, voice applications) as well as non-real-time services. Principal QoS controls include traffic classification, relative prioritization, queuing, bandwidth allocation, resource management, guaranteed throughput and ways to manage congestion. It is designed to minimize latency, jitter, loss rate and achieve high reliability.

QoS is not as serious an issue in Ethernet as it is in WLANs. Delivery of data around an Ethernet network does not require complex QoS mechanisms as modern Ethernet networks have a switched architecture unlike WLAN where the medium is shared. The end user would not notice the latencies much for sending/receiving emails and file transfers. But as WLANs become popular among business and public alike, the need for QoS has grown. Wired Ethernet has a large bandwidth (in the Gbps range), low packet error rates and packet overheads. But WLANs have a number of limitations. Stations contend with each other to access the medium, signal interference and signal attenuation with distance also take place. On WLANs the medium is shared, which means that the bandwidth available to each client is lower than that of Ethernet networks. Moreover, the packet-error rates and packet overheads are higher.

Another important factor is network congestion which is due to an increased number of clients attempting to access the medium and higher traffic volumes competing for bandwidth. All these characteristics can potentially limit the use of WLANs for delivering traffic for real-time applications such as voice and video applications. Without QoS guarantees applications often provide suboptimal responsiveness as a result.

There are many different types of multimedia applications present in WLAN, e.g. video on demand (VoD), voice over internet protocol (VoIP), IP Television (IPTV) etc. The characteristics of the data traffic for these applications are different from the traditional Ethernet traffic. VoD might require high bandwidth, low delay and guaranteed throughput, VoD is bursty in nature and requires strict limits on jitter and delay. If the QoS is compromised, the audio or video will be distorted. As a result, to ensure a high level of user experience, these traffic types need to be treated differently from that of traditional Internet traffic.

2.6 Challenges Associated with Video Streaming over WLANs

Video streaming over wireless network is an important and active area of research. Research areas include but are not limited to applications, content complexity, encoding configuration, streaming server, compression scheme, adaptation, client characteristics, quality assessment etc. Video is a frame based media. Video streaming can be defined as a server/client technology. Real-time streaming can be delivered by either peer-to peer (unicast) or broadcast (multicast). A main goal of streaming is that the stream should arrive and play out continuously with as small interruption as possible. Wireless links pose a significant challenge [56] for sending video streams. This is due to the fact that the wireless medium is shared and wireless links have low bit rate and high error rate compared to wired networks.

IEEE 802.11b and IEEE 802.11a/g WLANs operate with theoretical data rates of up to 11 Mbps and 54 Mbps respectively, but their effective throughputs are generally less than 7 Mbps and 25 Mbps respectively, depending on range and interference among various factors. Also different types of video formats and compression schemes (i.e. MPEG-2, MPEG-4, H.264, AVI, WMV etc.) affect the network differently. A typical SDTV stream consumes 2 to 4 Mbps, a DVD-quality stream consumes 8 to 10 Mbps, while a HDTV stream consumes approximately 18 to 24 Mbps using the MPEG-2 compression scheme [57]. As video distributions will invariably involve multiple streams, so an effective throughput of approximately 80 Mbps is required to enable delivery of 4 HDTV streams. The overhead associated with the wireless medium should also be added. Hence, it can be seen that current IEEE 802.11 WLANs are unable to provide sufficient bandwidth for high-quality video transport.

Also time-varying bandwidth, delay, jitter, and loss need to be addressed. Signal interference and attenuation with distance are a reality. Contention between stations, and background traffic are other factors that can cause effectively variable bandwidth for multimedia streaming applications. If the network becomes congested or the wireless channel fades, video tends to be affected much more than data. Net surfing, downloading a file, checking email etc. will be slower. But for video, if someone cannot watch uninterrupted broadcast quality video, the user experience will be unsatisfactory.

The bursty nature of video has important implications for the resource requirements of the network with regard to guaranteeing QoS. Moreover the type of network and transmission impairments experienced on the network will further complicate the task of guaranteeing of QoS. Real time video streaming requires different treatment from data traffic over WLAN. So it becomes important to manage the bandwidth carefully for video, voice, and data to achieve acceptable levels of QoS.

Video communication can be greatly facilitated by providing adequate QoS. The IEEE 802.11e standard specifies a number of QoS enhancement mechanisms at the MAC level. These IEEE 802.11e enhancements can be exploited to increase the WLAN throughput and decrease packet latencies. There are some subjective [58,59] (for evaluating QoE) and objective [60,61] (for evaluating

QoD) metrics available for evaluating quality of digital video. Examples are PSNR [62], VQM [63], PEVQ [64] etc. Every metrics has its advantages and disadvantages. Unfortunately at the moment none of them produce correlation to the exact Human Visual System (HVS) or human perception. This thesis deals with the QoD of video streaming over WLANs. The following section describes several video quality metrics.

2.6.1 Video Quality Metrics

There are broadly two, subjective and objective tests, ways to judge the quality of the received video.

2.6.1.1 Subjective Tests

Subjective testing is done to evaluate the QoE of the received video. Live trials are carried out where the human subject is the one who judges the quality of the video by rating the content. In this type of test the metric used is a Mean Opinion Score (MOS). Participants are requested to rate a clip from 1 to 5 being the highest quality. These values are then statistically analysed to obtain a MOS for the video. Subjective testing is time consuming, difficult to design and tends to be expensive. The viewing environment has to be optimised for the entire duration of the experiment.

2.6.1.2 Objective Tests

The objective evaluation techniques are mathematical models that emulate the subjective quality assessment results, based on criteria and metrics that can be measured objectively. They are classified as Full Reference Metrics (FR), No-Reference Metrics (NR), and Reduced Reference Metrics (RR) (Fig. 2.20).

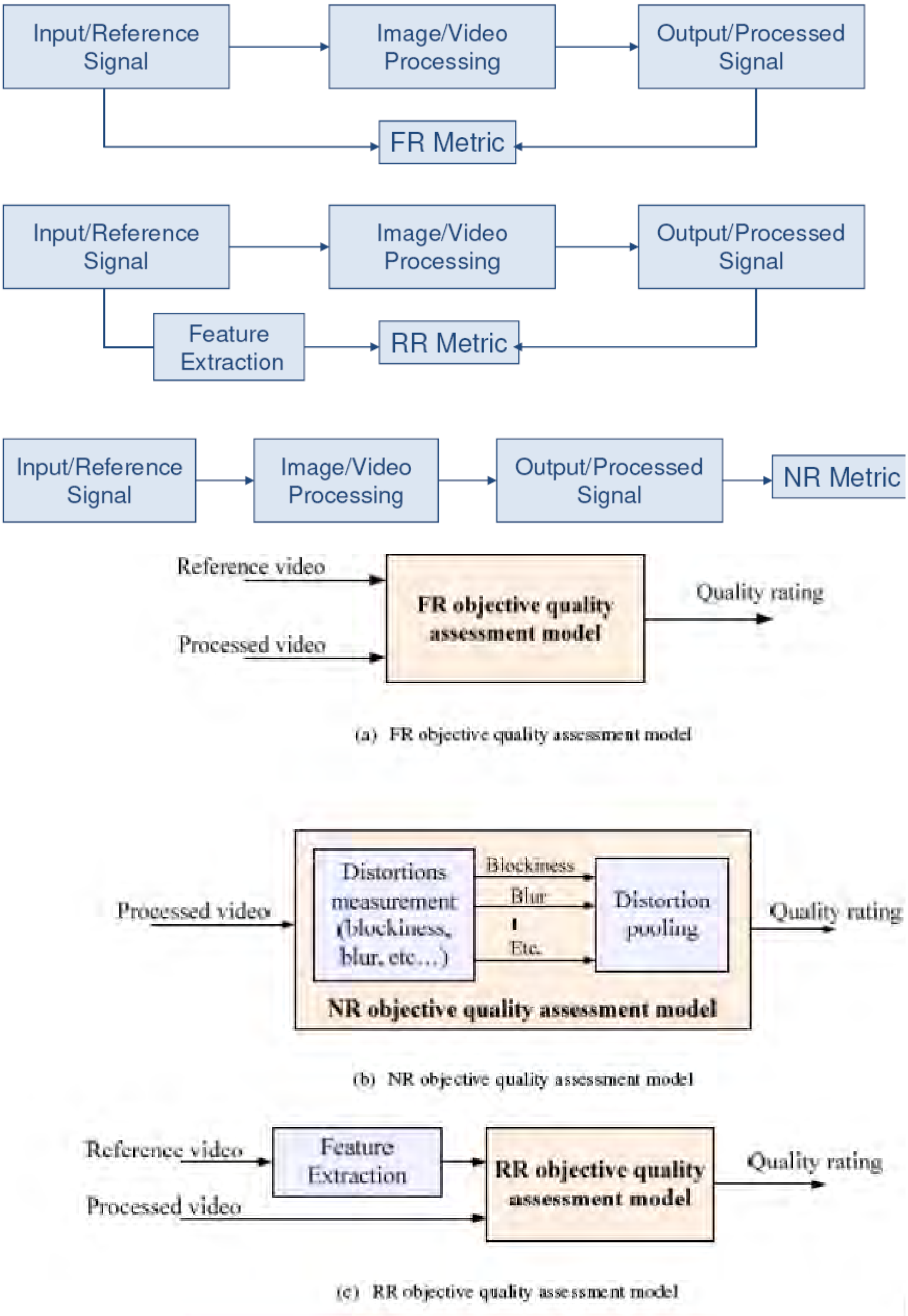

Fig. 2.20: Illustration of Different Types of Objective Video Quality Metrics (a) FR, (b) NR, and (c) RR

Both the host and client side videos are required for FR metrics. They perform a frame by frame comparison of the two video files to yield their result. In the NR method user's perception of a video stream is estimated without using an original stream as a reference. This makes them more flexible than FR metrics. In order to obtain accurate results NR metrics must be able to distinguish between image content and image distortion requiring complex processing. RR metric does not assume the complete availability of the host side reference signal and only partial reference information (e.g. motion and spatial details) is needed through a communications channel. Two well known metrics are described in the following section.

2.6.1.2.1 Peak Signal to Noise Ratio (*PSNR*)

Most of the FR quality assessment models share common error sensitivity based approach. An image or video signal whose quality is being evaluated can be thought of as a sum of a perfect reference signal and an error signal. It is assumed that the loss of quality is directly related to the strength of the error signal. Therefore, a natural way to assess the quality of an image is to quantify the error between the distorted signal and the reference signal.

This metric is simple in nature but has some limitations. Digital pixel values on which the calculation is performed, may not exactly represent the light stimulus entering the eye. The sensitivity of the Human Visual System (HVS) to the errors may be different for different types of errors. Two distorted image signals with the same amount of error energy may have very different types of errors

The most common objective metric is the Peak Signal to Noise Ratio (*PSNR*) metric. It is calculated on the luminance signal of a video file. This technique 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. 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 –

$$MSE = \frac{1}{mn}\sum_{i=0}^{m-1}\sum_{j=0}^{n-1}\left[I(i,j)-K(i,j)\right]^2 \quad \text{………..……………… (2.10)}$$

The *PSNR* is then calculated according to -

$$PSNR = 10\log_{10}\left(\frac{MAX^2{}_I}{MSE}\right) = 20\log_{10}\left(\frac{MAX_I}{\sqrt{MSE}}\right) \text{…..……… (2.11)}$$

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

2.6.1.2.2 Video Quality Metric (*VQM*)

VQM was developed by the National Telecommunications and Information Administration (NTIA). It was evaluated by the Video Quality Experts Group (VQEG) in their Phase II Full Reference Television (FRTV) test. It is a quadratic mapping scale for *PSNR* values. It is defined for *PSNR* values between 20 dB and 45 dB and converts PSNR values in the range into values between 0 and 1 [65]. 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. The *VQM* of a *PSNR* value is calculated according to

$$VQM(PSNR) = 0.0007816 \times PSNR^2 - 0.06953 \times PSNR + 1.5789 \text{ …….. (2.12)}$$

2.7 Multimedia and WLANs

Streaming multimedia is rapidly gaining in popularity and is predicted to be one of the highest revenue generating technologies in the near future. According to a study [35] from Insight Research, streaming multimedia could bring in $70 billion in revenue by 2013. It was forecasted that revenues from in-flight broadband entertainment would increase multiple times up from just under $7 million in 2009 [66]. The total investment in in-flight infrastructure is now half a billion dollars globally from 2009 through 2013. To re-energize new vehicle sales, automakers are planning to provide dynamic multimedia experience (i.e. integrating constantly connected in-vehicle infotainment systems) in the car via Wi-Fi/Bluetooth/GPS wireless technologies. Over 35 million in-vehicle infotainment (IVI) systems such as Fig 2.21 are expected to ship by 2015 [67]. Audi's Multi-Media Interface, BMW's iDrive, Ford's SYNC, Kia's UVO, Nissan's Leaf connected by AT&T USA, and Toyota's Entune are well known big names in this area.

Fig. 2.21: In-Vehicle Infotainment (courtesy: In-Stat)

Well known manufacturers like LG Electronics, Mitsubishi, Panasonic, Sharp, and Sony are working on wireless video technology as it represents the next generation in consumer electronic (CE) connectivity. With the current IEEE 802.11n and forthcoming IEEE 802.11ac and IEEE 802.11ad standards networks would be capable of transmitting high definition (HD) [68] and Ultra High Definition (UHD) [69] videos. The primary candidate technologies include: wireless home digital interface (WHDI), Wireless HD, and Wireless Gigabit (WiGig) etc. Triple-digit annual growth rate for high definition wireless video chips is forecasted for the next five years from 2010 onwards. In September 2010 [70], Quantenna Communications Inc., a HD-over-Wi-Fi startup raised $21 million to accelerate the deployment of 4 x 4 MIMO IEEE 802.11n Wi-Fi chipsets which would in turn allow the distribution of multiple high-definition (HD) video streams over WLANs with acceptable QoS.

A relatively new phenomenon of supporting rich media applications, e.g. video over WLANs, is to reduce costs of business. Many GSM and CDMA networks operators (e.g. O2 UK, BT, China Mobile, Softbank telecom in Japan, Verizon USA etc.) are considering offloading rich data on Wi-Fi networks which would address capacity crunch and congestion problems [71]. The goal is to utilise IEEE 802.11 networks for high bandwidth demand applications and the mobile network for other less demanding applications. It would save costs by diverting high volume multimedia traffic to an alternative access network and by creating a powerful, mass market engine that differentiates their offers and drives top-line growth (Fig. 2.22).

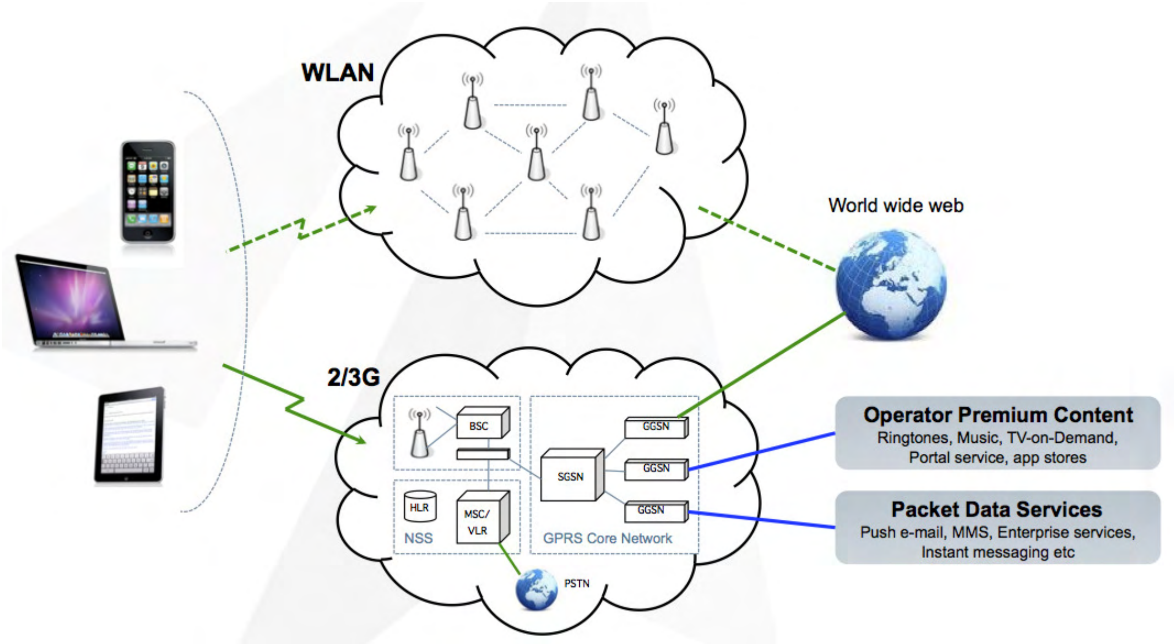

Fig. 2.22: Data Offloading to WLANs (Courtesy: Accuris Networks)

The extent of mobile data growth is widely reported. In 2010, O2 (The UK) reports that only 3% of its smartphone customers consume 36% of network bandwidth [72] by watching streamed video or playing online games. One study published in February 2010 by Allot Communications shows mobile data consumptions for three regions globally (Asia Pacific, EU, and USA) and goes on to show its breakdown by application [73]. HTTP downloads account for 19 per cent of worldwide mobile data, while browsing consumes 27 per cent and streaming (over HTTP) accounts for 29 per cent of the total. VoIP and IP niche applications, filling only three percent when combined leaving other applications to consume the remaining three percent. This is shown in Fig. 2.23. YouTube videos account for 32 per cent of the streamed total mobile data traffic. On March 15, 2011, Google reported that 35 hours' worth of video footage is now uploaded to YouTube every single minute from people all over the world [74].

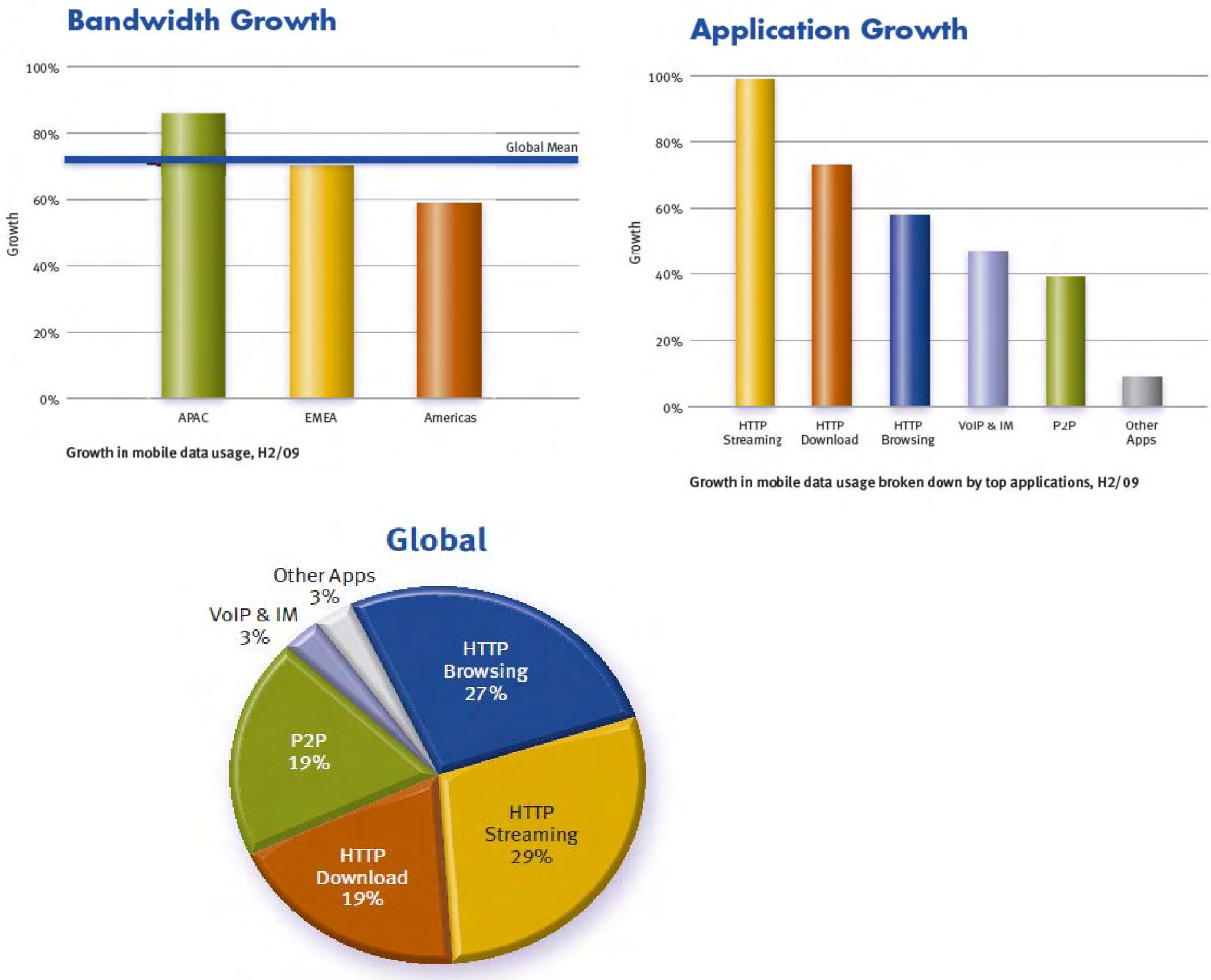

Fig. 2.23: Mobile Data Growth (Courtesy: Allot Mobile Trends)

So it is evident that the success of data-hungry smart-phones (e.g. the iPhone and other smartphones) and consumers' expectations for 'all-you-can-eat' data plans are leading to a mobile network capacity crunch [75]. O2 (UK) and AT&T network (USA) have scrapped unlimited data downloads for smartphone customers in 2010 [76,77]. Hence WLANs would be an excellent choice for GSM and CDMA networks operators for offloading rich multimedia content thereby addressing mobile network capacity crunch and congestion problems.

2.8 Applications

There are many applications for streaming video over wireless networks, for example:

- Video conferencing, Video on Demand, IPTV, Online gaming
- Border patrol / Homeland security
- Indoor cams / Outdoor cams / Home cams / Business cams
- Parking lots / Construction sites / Job site security
- Environmental monitoring e.g. wildlife watch
- Military / Commercial / Hobbyist / Personal
- Remote learning/ Education in institutions

With Video on Demand (VoD), video conferencing, and live multimedia streaming technologies educational institutions and companies can create, manage and broadcast high quality multimedia files to share with their web users such as students, academic staff, employees, customers etc. around the world and thus creating value for the enterprise by reducing cost and enhancing productivity.

The Wi-Fi Alliance [78,79] estimate that about 200 million households use Wi-Fi networks and there are about 750,000 Wi-Fi hotspots worldwide. Wi-Fi is used by over 700 million people and there are about 800 million new Wi-Fi devices every year. In-Stat [80,81,82,83,84], a multimedia market research company, predicts that the number of Wi-Fi enabled devices will continue to grow over the next five years, jumping from over 550 million in 2009 to nearly 1.7 billion in 2015. By 2015, over 800 million phones with embedded Wi-Fi are projected to ship. In 2012, Wi-Fi automotive shipments will reach nearly 20 million. Wi-Fi chipsets for notebook computers and mobile handsets are each expected to have revenue of over $1 billion in 2015. Although most Wi-Fi chipsets currently support the IEEE 802.11n standard, the IEEE 802.11ac and IEEE 802.11ad standards will eventually be the predominant technologies in near future. By 2015, 100% of mobile hotspot shipments will be IEEE 802.11ac enabled. The number of shipments of IEEE 802.11ac-enabled devices will reach nearly 1 billion by 2015.

Today many airlines offer VoD as in-flight entertainment to passengers via WLANs. Customers get the opportunity to select stored video or audio content and play it on demand. Big online game companies such as Blizzard Entertainment, NCsoft, Sony Online Entertainment etc. now generate high revenue from online games. DFC Intelligence, a market research and consulting firm, forecasts [85] that by 2012 the worldwide online game market will pass $13 billion.

Various organizations like government and/or private security agencies can monitor various properties, city facilities, parks, traffic intersections and important areas like airports, border regions in real time via live video streaming. Delivery of health care services and the quality of life can be improved. For example, doctors can assess patients remotely over the WLAN connections [86]. Hence this technology offers a horizon of unbounded possibilities.

2.9 Summary of the Chapter

This chapter presents the technical background of this thesis. Different video coding standards have emerged over time - the most popular being MPEG-1, MPEG-2, MPEG-4, H.264 etc. WLAN technologies are being increasingly used for multimedia transmissions. This thesis particularly deals with MPEG-4 video streaming over WLANs. There are different WLAN standards present in the market – while the most popular being IEEE 802.11b, IEEE 802.11a, IEEE 802.11g, IEEE 802.11e and IEEE 802.11n. The IEEE 802.11b/a/g standards provide a best effort service to all applications. They primarily use the DCF access method which provides an equal probability to each application in accessing the wireless medium. However, real-time services such as voice and video streaming require an upper bound to be imposed on the time required to transmit the packet. Hence, the IEEE 802.11e was developed to provide QoS to real-time multimedia applications.

The IEEE 802.11e standard describes two channel access mechanisms – namely EDCA and HCCA. HCCA guarantees reserved bandwidth for packets classified based on EDCA by using a central arbiter. EDCA defines four priority levels or access categories (ACs) for different types of packets. Each AC has four tuneable parameters - (*AIFSN*, CW_{min}, CW_{max}, *and TXOP*). ACs can be prioritized by tuning these four parameters which allows the higher-priority AC to win access to the wireless medium more frequently than the lower-priority AC.

Although the IEEE 802.11e standard introduced the concept of multiple ACs with four access parameters per AC, it does not describe how to implement a system for delivering QoS. In other words, IEEE 802.11e only provides for certain QoS enabling mechanisms – it does not in itself guarantee QoS. In a multimedia network, there may be multiple services, each with different QoS requirements. All of these should be supported in a cost-efficient manner by using network resources efficiently. So a significant challenge remains - how to employ the IEEE 802.11e mechanisms to support QoS for video streaming applications on WLANs. In the following chapters, solutions will be proposed backed by experimental analysis to guarantee performance improvement to deliver video over WLANs (in chapter 4). Based on the experimental analysis a novel QoS aware MPEG-4 video delivery algorithm will also be proposed and validated (in chapter 5) to address the performance of streamed video over IEEE 802.11b WLANs.

Chapter 3 LITERATURE REVIEW

WLAN is an active research area with considerable work carried out in the areas of Performance Analysis, QoS Provisioning, Admission Control, Voice/Video Streaming, Network Coding, Security etc. As this thesis is concerned with streaming video over WLAN, the literatures from the following research areas are described and are followed by a critical discussion -

- Performance Analysis
- QoS Provisioning
- Video Streaming
- Various Algorithms Proposed for Improving the Quality of the Streamed Video

After the discussion it would be evident that the IEEE 802.11b is not suitable for multimedia streaming and in the QoS enabled IEEE 802.11e standard there is no description of how to adjust the four access parameters (*AIFSN*, CW_{min}, CW_{max}, *TXOP*) associated with each access category. Proper tuning of the four IEEE 802.11e AC parameters to improve video performance over WLANs is still an open research question. In chapter 4 and 5 respectively, a QoS aware MPEG-4 video algorithm for streamed video over IEEE 802.11b WLANs is proposed and validated. Hence several algorithms proposed by other researchers are discussed in this regard in section 3.4. The algorithm presented in chapter 4 exploits the combined use of failed frame ReTx and GOPT in frame triplets (*PBB*) to minimise the probability of uncontrolled packet loss of the video streams at the expense of reduced quality thus achieving controlled and graceful video quality degradation under heavy network loads. Hence the proposed algorithm aims to deliver a QoS improvement by ensuring the realisation of the most favorable network conditions for the delivery of MPEG-4 video frames on WLANs.

3.1 Performance Analysis

Analysis of WLAN performance is an important area of research. Although WLANs have theoretical throughput limits associated with the corresponding standards, in reality the achievable limit is lower than that advertised. In their much cited work, Xiao and Rosdahl [87] have shown that due to the overhead associated with MAC mechanism, the IEEE 802.11 MAC displays a theoretical maximum throughput limit, implying that a straightforward increase in PHY bit rate will not necessarily lead to a corresponding increase in MAC layer throughput. Hence, an overhead reduction is required for IEEE 802.11 standards to achieve higher throughput. It is therefore necessary to develop MAC layer enhancements incorporating support for both QoS and higher throughput in order to facilitate the provisioning of current and emerging broadband multimedia applications.

Jun *et.al.* [88] have derived theoretical limits for MAC-level throughputs for various packet sizes on the IEEE 802.11 networks. They assumed that the networks have zero bit error and loss rate, there are always packets awaiting transmission, and the MAC layer does not use fragmentation. They showed that for a data rate of 11 Mbps, the maximum system throughput is approximately 6.1 Mbps for 1500 bytes packets.

Some important performance characteristics of the IEEE 802.11 DCF are reviewed in [89] by considering throughput, fairness, and delay for the IEEE 802.11b and IEEE 802.11g enhancements. Fixed overhead (*DIFS, SIFS* etc.), increasing contention between stations (which results in collisions), and transmission errors are investigated. It is shown that under imperfect channel conditions switching to a lower bit rate is beneficial if the frame error rate exceeds some significant threshold. But stations switching to lower bit rates to adapt to bad channel conditions may significantly lower the throughput of stations that use higher bit rates. The paper suggests that to improve short-term fairness, an optimal DCF-like access method needs to use the equal size contention window for all contending stations. Long-term fairness of the IEEE 802.11 DCF results in significant unfairness at the level of TCP connections, because the access point does not benefit from sufficient capacity to convey download traffic.

3.1.1 Discussion

These studies have helped us to gain a good insight into the practical MAC throughput for WLANs. The important findings observed in [70] ultimately led to the latest IEEE 802.11n standard which recommends changes to both the PHY and the MAC layers for improving QoS. The upper limit of 6.1 Mbps system throughput (for 1500 bytes packets) described in [71] was used as a guide in designing our experiments described in chapter 4 and 5. In [72], the time delay introduced in the network for the random backoff is not included in the throughput and other calculations and so somewhat higher values for different entities are calculated. As an example, through numerical calculations it is shown that for IEEE 802.11b and IEEE 802.11g, the efficiency at 54 Mbps and 11 Mbps become 0.7 and 0.6 respectively. Thus, a single station sending frames of 1500 bytes over IEEE 802.11b can at most obtain throughputs of 8.69 Mbps and 37.26 Mbps over IEEE 802.11g. Nevertheless the conclusions are insightful. [90,91,92,93,94] are also recommended for further reading about performance analysis of WLANs.

3.2 WLAN Performance Enhancements (EDCA Perspective)

In [95], Medepalli and Tobagi developed an analytical model of IEEE 802.11 WLANs from a fixed point analysis perspective rather than the traditional Markovian analysis. The analytical model can accommodate arbitrary topologies along with directional antennas, multiple channels as well as different per node traffic requirements. Here different traffic flows entering the network are assumed to be independent Poisson processes (at a packet time-scale). There is no restriction on packet size statistics. They claimed that such an approach works well when the number of users is relatively large and when metrics such as throughput or delay are being considered. Using their model they showed that CW_{min} offers a far greater control over the throughput and adapting the initial contention window CW_{min} is more beneficial than adapting to a value between fixed CW_{min} and $CW_{max.}$

Aad and Castelluccia [96] proposed a priority scheme by differentiating interframe spaces *(IFS)*. Veres *et.al.* [97] presented priority schemes by differentiating the initial backoff window size and the maximum window size. In [98] an adaptive algorithm was proposed to dynamically re-calculate the CW_{min} value accordingly to the specific traffic class and changes in the network load. Scalia and Tinnirello [99] developed an IEEE 802.11e MAC simulator using *C++* to analyze the behaviour of IEEE 802.11e differentiation mechanisms in presence of data and multimedia traffic. They show that to optimize the overall system performance both *AIFSN* and CW_{min} values need to be adjusted. Raimondi and Davis [100] concluded that a mechanism that incorporated both

AIFSN and CW_{min} exhibited a better performance than a mechanism based solely on either parameter. A Class Based Differentiated Service (*CBDS*) scheme was then developed which described three service classes - gold, silver, and bronze for different classes of service.

Pong and Moors [101] showed that by using judicious CW and *TXOP* values, target latency and throughput performance can be obtained. Andreadis and Zambon [102] proposed an algorithm for dynamic *TXOP (DTXOP)* assignment. *DTXOP* is periodically updated according to the traffic conditions of each Access Category. Through simulation they showed that the proposed *DXOP* allows to maintain fairness between upstream and downstream channel access times and to enhance delay and throughput performance. Through simulation Suzuki *et.al.* [103] showed that *TXOP* parameter can improve the audio and video quality in the presence of transmission errors. The average video delay, loss ratio, and media synchronization quality are improved in the downlink direction along with user level QoS.

To provide acceptable QoS, in [104] Xiao *et.al.* proposed a two-level protection and guarantee mechanism for voice and video traffic in IEEE 802.11e Wireless LANs. In the first-level, the existing voice and video flows are protected from the new and other existing voice and video flows. In the second-level protection, the voice and video flows are protected from the data traffic by tuning the CW_{min} and *AIFSN* parameters. Simulation results show that the proposed mechanism is effective in terms of facilitating multimedia traffic and improving the utilization in the channel capacity.

In [105], two admission control based bandwidth partition schemes (called static and dynamic partitioning) for multimedia (data/voice/video) traffic over IEEE 802.11e WLANs were proposed and analysed. As the names suggest, available bandwidth is allocated in static and dynamic manners among various traffic types based on the current voice/video/data traffic load condition. Performance metrics used to indicate quality were average throughput per voice/video/data flow, total throughput, number of accepted and active flows, transmission time etc. Simulation results show that the Dynamic Scheme is better than the Static scheme. Unfortunately constant bit rate (CBR) video traffic is used for simulations and not real life video which is in general variable bit rate (VBR).

Through OPNET modelling, Sebastião and Correia studied [106] the effect of tuning various AC parameters for six different services - VoIP, Video Streaming, Video Telephony, HTTP, FTP and Email. They argue that the *AIFSN* value is the best way to separate traffic, (especially for the Real Time streams). For the lower priority traffic classes (data), it is best to change the CW_{min} and CW_{max}. *TXOP* can be used to increase the maximum achieved throughput for given traffic class, but it will increase overall delay, thus is only recommended for data-centric networks with few voice users.

A scheme is proposed in [107] to ensure both intra and inter QoS differentiation in IEEE 802.11e WLANs. Each traffic class monitors the MAC queue and computes at runtime the *TXOP* value based on the queue length. An admission control function is also introduced to protect the admitted flows and maintain the network in steady state. However, this scheme does not consider the frames that may arrive during the transmission. These frames also need to be transmitted using current transmission opportunity.

In [108], a scheme called *adaptive transmission opportunity* (*ATXOP*) is proposed to address the unfairness problem in the IEEE 802.11e networks adopting the multi-rate scheme. The unfairness in terms of throughput arises due to the time varying data transmission rate in multi-rate IEEE 802.11e networks. To solve the problem, it is proposed to assign stations of lower data rate larger *TXOP* and stations of higher data rate smaller *TXOP*. At first an average transmission rate of all the stations in the network is calculated. Then the current transmission rate of each station to the current average rate is compared. If it is found lower or higher than the average rate, the *TXOP* will be changed by a factor related to the ratio between its current transmission rate and average transmission rate of the network. Although the unfairness issue which arises due to the time varying data transmission rate in variable rate WLANs is addressed, the allocation of *TXOP* for various data/multimedia traffics is not properly analysed.

3.2.1 Discussion

There have been numerous performance studies of the IEEE 802.11 DCF and IEEE 802.11e EDCA under saturated [109, 110, 111, 112, 113] and unsaturated [114, 115, 116, 117] channel conditions. Most of the analytical methods are based on a multidimensional discrete time Markov chain. Others have considered queuing theory to analyze these mechanisms [118].

From the studies mentioned earlier it can be concluded that DCF is not suitable for multimedia applications that have certain QoS requirements. As DCF treats all traffic types be it data or multimedia in the same way, i.e. without any priority, a station with real-time multimedia traffic may have to wait for a long period of time to send packets. As a consequence, real-time applications can suffer from a poor performance under DCF operation.

The IEEE proposed the enhanced distributed channel access *(*EDCA*)* MAC mechanism to support prioritized channel access for real time multimedia streams in order to realise an acceptable QoS. But the standard does not describe how to guarantee strict QoS required for real-time services. A number of mechanisms which include admission control, rate control, proper tuning of the IEEE 802.11e parameters *(AIFSN,* CW_{min} , CW_{max}*, TXOP)* etc. have been proposed [119, 120, 121] by researchers to enhance network performance and thus to achieve acceptable QoS. Some of the relevant published works are described in the earlier section.

Different studies [79, 80, 84, 85] suggested tuning of the IEEE 802.11e AC parameters *(AIFSN,* CW_{min}, CW_{max}*, TXOP)* separately or in a combination should be employed to achieve acceptable QoS. But there are very few papers available which describe strategies involving all the four IEEE 802.11e parameters *(AIFSN,* CW_{min}, CW_{max}*, TXOP)* plus admission control. Also the relative importance of the parameters, i.e. which parameter is most effective in tuning to achieve higher throughput, less delay for video streaming has not been reported. It can be concluded after reviewing these studies that they agree in general that *by tuning the parameters network performance can be enhanced.* This conclusion guided our work as in our experiments (described in chapter 4) these parameters were varied over time for optimum network performance in the context of streaming video over WLAN. The results obtained in chapter 4 helped us to design and validate a QoS aware MPEG-4 video delivery algorithm for streaming video over IEEE 802.11b WLANs based upon GOP truncation and failed frame retransmission (described in chapter 5).

3.3 Video Streaming over WLANs

In [122], an adaptive system for improving videophone transmission over IEEE 802.11e is proposed to address the AP bottleneck issue and the problem of adjusting video source rate to improve the network performance. The AP bottleneck issue is dealt with by prioritising the AP in terms of the transmission opportunity (*TXOP*) values. The second issue is addressed by guaranteeing the voice traffic throughput to some extent while transmitting as much video traffic as possible. Voice codec G.711 and video codec H.263+ (300 frame GOP, 1 *I* frame and 299 *P* frames) have been used for NS-2 simulation. Simulation results show that the proposed system can improve the number of videophone sessions from 7 to 10.

Through Network Simulator 2 (NS-2) simulation and experimental tests, [123] assesses the MPEG-4 video streaming performance over the IEEE 802.11e WLANs by tuning the *AIFSN* and CW_{min} parameters separately. Subjective testing (PSNR based VQM metric) was also employed to report end user satisfaction of the different types of streamed videos. The videos were encoded at different rates ranging from 100 kbps to 1000 kbps with a step of 100 kbps. *AIFSN* and CW_{min} parameters were varied from 4 to 21 (step size 1) and 10 to 60 (step size 5) respectively. According to the 802.11 standard CW_{min} values should be in the order of 31, 63, 255, 511…1023. (i.e. $2^n -1$, n is a nonzero integer). Hence the choice of CW_{min} values in the paper is not optimal. It is shown that the relation between video quality and the CW_{min} or *AIFSN* parameter is highly non-linear. Different videos were affected differently by the same level of CW_{min} or *AIFSN* values.

Koucheryavy *et.al.* [124] show that best effort IEEE 802.11b WLANs are not capable of delivering multimedia services such as live video streaming efficiently. They describe results for various signal-to-noise ratios *(SNRs)* and competing TCP and UDP traffic volumes. Shimakawa *et.al.* [125] carried out a simulation based study concerning a WLAN's ability of supporting video-conferencing and data applications. They also showed that the use of EDCA enhances performance of both MPEG-4 video and data applications compared to DCF.

Demircin and Beek [126] used the NS-2 simulation tool to analyse their proposed bandwidth estimation technique which operates by adjusting the video bit-rate dynamically and includes a delay-constrained rate adaptation algorithm at the sender for the IEEE 802.11e standard. They show that streaming with bandwidth estimation and rate adaptation achieves a higher PSNR gain compared to streaming without rate adaptation.

Kuang and Williamson [127] experimentally studied the performance of multimedia (using the Real Audio and Real Video applications) streaming over IEEE 802.11b wireless LAN under different channel error conditions and considered the effect of competing TCP/IP traffic on the quality of UDP-based Real Media streaming sessions. The maximum measured throughput was 4.6 Mbps for 'excellent' (i.e. signal strength > 75%) channel conditions. Also it was demonstrated that competing TCP/IP traffic had little impact on streaming. Through experimentation, Gopal *et. al.* [128] demonstrated that streaming multiple MPEG-4 AVC encoded video clips over best effort IEEE 802.11b network face a significant bottleneck due to MAC and physical layer overheads in DCF mode. They derived theoretical maximum application level throughputs for various packet sizes and showed that the network is affected differently by different packet sizes. Also the effect of channel load and receiver location on packet losses for video traffic was studied.

Two packet mapping schemes are compared to two packet dropping schemes in the context of video streaming over IEEE 802.11e WLANs in [129]. The packet mapping schemes determine which IEEE 802.11e EDCA queue each video packet is sent to. One mapping scheme treats all packets equally, while the other differentiates the video packets of different slice types (*I, P,* and *B*) according to their priority. The packet dropping schemes aim to avoid MAC layer congestion by dropping some of the packets before arriving at the MAC. One packet dropping scheme applies an even amount of loss to each video stream while the other applies an even statistic which results in an uneven amount of loss to each video stream. Using NS-2 simulator and J.144 video quality estimation tool (with buffer queue length 600, H.264 video standard, video resolution 720 x 576, video frame rate 25 fps, packet size on average 1290 byte, three video files with different spatial and temporal characteristics) the authors show that the schemes that differentiate video packets allow for a more gradual video quality degradation. At the same time the packet dropping schemes offer much better delay performance compared to the packet mapping schemes as they drop packets in order to avoid congestion.

An alternative backoff mechanism was proposed in [130] for QoS provisioning of video streaming in wireless IEEE 802.11 home networks. The method combines the advantages of both the reservation based methods and the contention based accesses. Analytical and simulation results show that this solution can improve the system performance in terms of network throughput and delay, hence enabling effective QoS support. An AIMD based algorithm is then proposed to enable efficient resource allocation for video streaming over wireless LANs.

In [131], a content based perceptual quality reference-free metric for various wireless MPEG-4 video streaming applications is proposed and analysed using FFMPEG analyser and NS-2 simulator. Cluster analysis is used to classify different contents into three specific content types of ‘slow movement’, ‘gentle walking’, and ‘rapid movement’ based on the spatial and temporal feature extraction. QCIF resolution (176 x 144), 10 to 30 fps, 9 frame GOP videos were chosen for this work. However animation type content, higher resolution and full GOP videos were not included which would give some interesting insight regarding the proposed metric.

In [132], several packet mapping schemes were compared for sending concurrent H.264 video streams over IEEE 802.11e WLANs. These mapping schemes incorporate the video server to assign packets with the correct priorities to IEEE 802.11e queues. Analyses of the NS-2 simulation results show that the various mapping schemes used produce different types of video impairments, i.e. different mapping schemes exhibit different loss patterns in the video sequences. The severity of the impact that these impairments have on video quality is content dependent. Subjective video quality tests were undertaken to assess the end users' judgments of the various types of impairments produced from each mapping scheme.

In [133], the authors advocate the use of packet level Forward Error Correction (FEC) mechanisms to improve video multicast performance. They take a cross layered approach and use MadWifi driver for the Atheros chipsets for the testbed. The analyse the performance of the network by using comparing PSNR values of a video clip of 352 x 288 resolution under various transmission rate and FEC scenarios for IEEE 802.11b WLANs. They conclude that the Packet Error Rate (PER) increases exponentially with distance and using a higher transmission rate together with stronger FEC is more efficient than using a lower transmission rate with weaker FEC for video multicast.

A scheme was proposed for enhancing QoE of Audio-Video IP transmission at the receiver in [134], which utilizes the QoE tradeoff relation between spatial and temporal quality caused by error concealment and frame skipping for H.264/MPEG–4 AVC. The study was carried out for different types of video contents containing *I* and *P* frames only (i.e. no *B* frames were employed in the video files).

QoE was estimated [135] for six multimedia contents using NS-2 simulation over a WLAN by the method of successive categories, which is a psychometric method. Multiple regression lines were used to perform QoS mapping between the MAC-level and user-level. The regression lines estimated QoE from MAC-level QoS. Also the effect of content types on the QoE by using estimated values from MAC-level QoS was evaluated.

[136] evaluates the performance of in-flight video streaming over IEEE 802.11n with respect to the frame aggregation schemes considering the QoS requirements of in-flight video using NS-2 simulation. The results exhibit that the A-MPDU (Aggregate MAC Protocol Data Unit) frame aggregation scheme achieves a higher throughput than the A-MSDU (Aggregate MAC Service Data Unit) frame aggregation scheme in both ideal and error prone channel conditions. The simulation also reveals that the number of in-flight entertainment devices which can be served by an IEEE 802.11n access point for a packet size of 1400 bytes is 39. So it can be concluded that for a typical short haul aircraft (with 100 passengers) approximately 3 IEEE 802.11n access points which operate on non overlapping channels are needed for supporting wireless in-flight video.

[137] proposes a scheme called Instantaneous Multiple-Receiver Frame Aggregation (IMA) based on the concept of congestion triggered aggregation for HD video streaming over IEEE 802.11n. Through Qualnet simulation the authors show that their IMA scheme outperforms the traditional IEEE 802.11n aggregation in terms of video stream throughput, delay, jitter, and loss. They conclude that the number of video streams that can be supported on the IEEE 802.11n networks depends heavily on how the frame aggregation is implemented. Simulation results show that for 12Mbps video, traditional IEEE 802.11n and IMA enabled IEEE 802.11n can carry 3 and 7 streams respectively

on a 135 Mbps PHY. For 18 Mbps video the number of supported video streams are 2 (IMA) and 5 (IEEE 802.11n) respectively.

In [138], an analytical model is developed for the performance study of an IEEE 802.11n WLANs to support voice and video applications in IEEE 802.11n. The paper shows that IEEE 802.11n's enhanced MAC mechanisms (e.g. frame aggregation and bidirectional transmission) can effectively improve the network capacity by not only reducing the protocol overheads but also smoothing the AP-bottleneck effect. Voice and video capacity under various MAC mechanisms are compared as well.

3.3.1 Discussion

The majority of the papers published in the literature regarding streaming video over WLANs are simulation based. Most often OPNET, NS-2 etc, simulation tools are used for this purpose. Although simulation based studies are convenient, in reality no simulation tool could ever emulate all the aspects of a real life network. Most of the published work involving MPEG-4 video streaming over WLANs treat video as an aggregate stream, i.e. do not consider the inherent *IPB* frame and GOP based nature. The few ones that consider the *IPB* frame based nature of MPEG-4 video do not always consider all three frame types. Some papers use arbitrary CW_{min} values and GOP length (MPEG-4 standard defines a GOP length of 15 for PAL systems) for simulations. According to the IEEE 802.11 standard CW_{min} values should be in the order of 31, 63, 255, 511…1023 (i.e. 2^n -1, n is a non-zero integer). In contrast to this trend, this thesis describes experimental studies involving real MPEG-4 video content streamed over real IEEE 802.11b/e WLANs in chapter 4. The choice of

AIFSN, CW_{min}, *TXOP* values used in the experiments described in this work always followed the IEEE 802.11 standard. [139,140,141,142,143,144] are also recommended for further reading about multimedia performance.

3.4 Algorithms Related to Video Streaming over WLANs

Over the last couple of years there have been several algorithms suggested by various researchers to improve streamed video quality over WLANs. As our proposed QoS Delivery Algorithm (described in detail in chapter 4) is based on novel intelligent packet dropping (i.e. truncating GOP frame triplets in order of their importance to address the network delivery of video) and MAC level failed frame retransmission, only the relevant schemes are discussed here.

A novel buffer underflow avoidance scheme for multiple-source multimedia delivery is proposed in [145] which is based on the dynamic buffer occupancy estimation for highly loaded network conditions during content delivery. In order to overcome varying network conditions, a double buffering architecture is employed which uses virtual multiple buffers associated with multiple network connections with the classic decoding/playing buffer. The scheme balances the streamed video between the multiple connections enabling to achieve high quality without content adaptation to network conditions. Through NS-2 simulation and using PSNR metric based on frame loss and throughput of the streamed video with increasing number of users, it is shown that that the scheme performs well compared to other solutions. For testing purposes five MPEG-2 encoded videos were used with frame rate of 25 fps and GOP lengths of 12 (*IBBPBBPBBPBB)*. It would be interesting to see how their scheme performs for MPEG-4/ H .264 videos with standard 15 frame GOPs.

Chen *et.el.* describe [146] a cross layer content-aware retry limit adaptation (CA-RLA) scheme for streamed video over WLANs that dynamically adapts the retry limit for each video packet based on its loss impact. Here compressed video stream is either pre-stored in the server or sent to the server through an access network. In the off-line encoding process, the encoder estimates the amount of error propagation caused by each packet if it is lost during transmission. The proposed scheme increases the retry limits of packets of higher loss impacts, while reducing the retry limits of packets of lower loss impacts so as to minimize the overall error propagation in a GOP under the delay constraint of video presentation. Using the OPNET network simulator for IEEE 802.11b networks and taking PSNR metric into consideration the authors show that the proposed adaptation scheme can mitigate the error propagation due to packet loss and assure the on-time arrival of packets for presentation thus improving the video quality. For experimental purposes three 300-frame QCIF (176 × 144) test sequences were used which were pre-encoded at 30 frames/s and 384 kbps with GOP size of 30 frames. But higher resolution videos (e.g. SD or HD) and a MPEG -4 GOP size of 15 would be more interesting for simulation as that video resolution (SD/HD) would test this algorithm more rigorously for the IEEE 802.11b networks.

An on-off queue control mechanism (OOQC) is proposed in [147]. The scheme involves controlling the number of active nodes on the channel in order to reduce collisions under heavy traffic conditions by source rate adjustment. A low priority early drop (LPED) method is also employed to drop the packets at the queue according to packet relative priority index (RPI) provided by scalable video coding. It is argued that the mechanism maintains high network throughput while keeping packet loss due to collision as low as possible. Video quality is measured at the AP for the proposed OOQC mechanism through NS-2 simulation and PSNR metric. The maximum packet size, the GOP length and video resolution are set to 500 bytes, 8 frames and CIF sequence respectively for simulation. Simulation results show that the proposed OOQC scheme outperforms EDCA in received video quality. Larger packet size (e.g. 1500 bytes), a 15 frame GOP and higher resolution video would have provided more relevant results about the performance of the proposed scheme.

A cross-layer based video transmission architecture is described in [148]. The architecture consists of an application layer, a transport layer, and MAC (Medium Access Control) layer. The architecture is based on the priority of MPEG-4 video frames and the mechanism adaptively controls the transmission rate by dropping the frames based on bandwidth estimation. Through 100 second long NS-2 simulations, the scheme is demonstrated (using PSNR metric) to improve the end-to-end streamed video quality. In this work the max retry limit chosen for *I, P, B* frames are 8, 8, and 4 respectively. There is no satisfactory explanation for choosing these values. In a typical MPEG-4 GOP (for PAL systems) there are 15 frames (1 *I*, 4 *P* and 10 *B*) with the priority being *I* > *P* > *B.* The loss of the only *I frame* would translate into not being able to decode the remaining 14 *P* and *B* frames at the client as these frames depend directly or indirectly on the successful transmission of the *I* frame. If an *I* frame is lost but the remaining *P, B* frames are transmitted, this would waste network resources and add to the total network delay. Hence, *I* and *P* frames should not get equal retransmission opportunities. Also this scheme at first throws away the *B* frames one by one based on the transmission rate. Then *P* frames are discarded. The typical MPEG-4 GOP is *IBB PBB PBB PBB PBB.* There is an interdependency of the frames as described in chapter 2. If we were to discard the last two *B* frames, there is no point in keeping the last *P* frame and so on. So this algorithm does not optimise bandwidth usage. Hence it would be more sensible to throw away entire triplets instead if the available bandwidth becomes scarce which has been implemented in the QoS aware MPEG-4 video algorithm described in chapter 4 and 5 in this thesis. Also a 100 second long simulation, to our opinion, is of too short a duration. As time proceeds more and more frames arrive at the buffer and more accurate analysis of throughput, delay, loss rate can be performed.

In [149], a cross-layer time-based retransmission scheme is proposed and realised for the WLANs (contrary to the default count-based scheme as defined by the IEEE 802.11 standard) to provide QoS for delay sensitive video streaming applications. According to the proposed architecture, the retransmission deadline is assigned by the application layer according to the application's specific requirements for the transmitted media data. Subsequently the MAC layer dynamically determines whether to send or discard a packet based on a retransmission deadline attached by the video server. It is argued that this can significantly reduce the number of late packets. In addition, the proposed mechanism can provide differentiated error protection to different types of MPEG-4/H.263 video packets. OPNET simulation and experiments with LINUX based MADWifi driver were carried out for IEEE 802.11b WLANs. It is concluded that the time based retry approach outperforms the count-based retransmission mechanism in terms of video quality.

[150] elaborates a scheme for error protection of video streaming over WLANs. It describes the impact of frame retransmission on video QoS by taking a two tier strategy. At first the scheme calculates the loss impact of *I* and *P* frames but not *B* frames of the streamed video in an off-line coding process. Then based on the estimated loss-impact values, a ranked prioritized retransmission scheme, called "Greedy Algorithm" is proposed. The clients determine whether or not to request a retransmission for a lost packet according to its play out deadline. If the server receives a retransmission request for a lost packet, it would use the rank of the packet's loss-impact value to choose the one with the larger loss-impact value to transmit from either of this lost packet and the regular packet(s) with a similar total size, and drop the other. Frame by frame PSNR comparison for 5% and 10% packet loss rates was used to show the performance improvement of streamed video. It is argued that the packet loss within a *B* frame won't result in any error propagation. In reality loss of every frame (*I, P, and B*) results in quality degradation for video although a *B* frame has the least impact on the video quality of all the three frames. Also the later *P* frames are less important than the earlier *P* frames. But the paper does not cater for that. At the same time, using a GOP with size of 30 (MPEG-4 standard defines a 15 and 18 frame GOP for PAL and NTSC systems respectively) with a GOP sequence of *IPPPPP*… (instead of *IBBPBBPBB*….) does not reflect the actual MPEG-4 standard. This is the paper that is somewhat related to our proposed QoS aware MPEG-4 video delivery algorithm but in our work all the relevant parameters and their values reflect the MPEG-4 standard in contrast to this work.

3.4.1 Discussion

The Peak Signal to Noise Ratio (PSNR) metric has been used to measure quality in several papers referenced above although this has, however, been shown to be a poor indicator of quality [151]. Some researchers suggest that the time based retry approach outperforms the count-based retransmission mechanism in terms of video quality although most of the papers found in literature rely to count-based frame retransmissions.

Different papers suggest different methods to improve video QoS over WLANs. Some algorithms work on a single OSI layer while others take a cross layered approach. For implementation simulation tools such as OPNET, NS-2 have been used. Different simulation tools would not produce the exact same result for even the same algorithm with similar settings. This is a dilemma faced by researchers around the globe. It is also not easy to compare the performance of the presented algorithms to one another as they use different video formats, resolutions, frame rate, frame size. Also the test video files used in the simulations or experiments would have different spatial and temporal characteristics.

As discussed in chapter 2, the MPEG-4 video standard uses a GOP length of 15 (*IBB PBB PBB PBB PBB*) and a frame rate of 25 fps. But different research groups use different GOP lengths and video frame rates other than the standard value. It was noticed that GOP length values between 8 and 300 were used for simulation settings in various simulation tools such as OPNET, NS-2 etc. Many papers perform simulations with *I* and *P* frames only, i.e. discarding *B* frames, by arguing that the effect of loss of *B* frames is negligible.

However if a simulation is run for the entire duration of a MPEG-4 Hollywood movie (of typical duration 100-150 minutes) then the frame loss for *B* frames would not be miniscule and hence could not be discarded in the analysis. In this case the neglecting of *B* frame loss rate would be somewhat far from reality.

Based on the above discussion and the knowledge gained from chapter 4 regarding the influence of the *IPB* frame based nature of the MPEG-4 video, a novel QoS aware MPEG-4 video delivery algorithm is proposed and validated through extensive simulation for both uplink and downlink video traffics using a computer model written in the *C* programming language in chapter 4 and 5. The test videos were MPEG-4 compliant (i.e. frame rate 25 fps, GOP length 15, GOP pattern *IBB PBB PBB PBB PBB*, Display Resolution 720 x 576 pixels, *B* frame frequency 2).

It employs two mechanisms namely frame retransmission (ReTx) and GOP truncation (GOPT). The GOPT mechanism proposed here is novel and unique and involves in selectively dropping frame triplets in order of their importance from the GOP to reduce the number of video packets required to be transmitted to address the scarcity of network resources. The GOPT mechanism reduces the probability of buffer overflow at the expense of a reduced QoE by discarding the comparatively less important video frames in triplets (*PBB*). The count based ReTx mechanism is focused on minimizing packet loss due to MAC collisions in the WLANs. It effectively increases the QoS by minimizing the transmission losses at the expense of an increased buffer overflow probability.

The proposed QoS aware MPEG-4 video delivery algorithm aims to replace uncontrolled packet loss due to buffer overflow and MAC collisions by a controlled prioritized packet loss scheme that permits a graceful degradation in QoD for MPEG-4 video streamed over IEEE 802.11b networks. To the best of our knowledge, this is the first time where a solution has been proposed and validated for enhancing the quality of streamed video over IEEE 802.11b WLANs by breaking up the MPEG-4 video into its constitute frames and then by combining the ReTX and GOPT mechanisms to minimize frame losses and eliminate potentially catastrophic buffer overflow .

3.5 Summary of the Chapter

Previous studies have shown that IEEE 802.11b is not suitable for multimedia streaming. The IEEE802.11e standard was introduced as an enhancement to IEEE 802.11b for addressing QoS concerns for multimedia streaming over WLANs. The IEEE 802.11e standard defines four parameters *(AIFSN,* CW_{min}, CW_{max}*, and TXOP)* which can be tuned to improve network performance. The EDCA mechanism is only a QoS enabling mechanism – it can be used to support QoS, it does not in itself provide QoS. It needs to be incorporated into a QoS provisioning framework/system. Hence numerous papers have been published about tuning these four parameters for data, voice, and video streaming – where the majority of the studies have been performed through simulation. To date there is no report on the standardized relative prioritization of the parameters, i.e. among the four parameters which one is the most effective for different types of traffics. The conventional use of IEEE 802.11e still has not delivered the required performance improvements to deliver video. Video over WLANs is an emerging area of research. Most of the research papers treat video as an aggregate traffic stream and do not consider the characteristic *I, P, B* frame based nature of video in context of WLAN. In our work, a novel approach has been adopted where a differentiated service to the individual constituent *I, P, B* video frame types was provided to enhance system throughput. Also all four Access Categories *(AC_VI, AC_VO, AC_BE, AC_BK)* available under the IEEE 802.11e standard were employed and assigned different levels of prioritization as described in appendix B. Hence this thesis proposes an alternative use of QoS mechanisms provided for under the IEEE 802.11e standard that can enhance the network performance to deliver video.

The work presented in appendix B is experimental in nature which represents real world scenarios for video streaming over WLANs as opposed to simulation or analytical studies. Based on these results, a unique QoS aware MPEG-4 video delivery algorithm is proposed, implemented and validated in chapter 4 and 5 that improves MPEG-4 video QoD over WLANs by exploiting the *IPB* frame based nature of MPEG-4 video. It exploits the inherent coupling of two mechanisms, namely failed frame ReTx and GOPT to achieve the ITU-T target specified for loss rate ($\leq 1\%$) of streamed video transmission.

Chapter 4 A Novel QoS-Aware MPEG-4 Video Delivery Algorithm over the Lossy IEEE 802.11b WLANs

4.1 Introduction

A novel QoS aware MPEG-4 video delivery algorithm is proposed in this chapter that improves MPEG-4 video QoD over WLANs by exploiting the *IPB* frame based nature of MPEG-4 video. It will be implemented and validated in chapter 5.

Before proposing and implementing the QoS aware MPEG-4 video delivery algorithm different experimental scenarios were investigated and analysed. Mainly four scenarios were considered in terms of analyzing video QoD streamed on IEEE 802.11 WLANs -

a) Comparison of Wired versus Wireless Video Streaming over IEEE 802.11b WLANs

b) Effects of Background Traffic Loads on Streamed Video over IEEE 802.11b WLANs

c) The Effects of Contention between Stations on Video Streaming Applications *d) Investigation of the Impact of TXOP on Parallel Multimedia Streams over QoS Enabled WLANs*

They are described in detail in the appendix -B section. The proposed QoS aware MPEG-4 video delivery algorithm is based on the conclusions achieved from the above tests that individual constituent *I, P, B* MPEG-4 video frame types can be exploited to enhance the performance of the network delivery of MPEG-4 video.

4.1.1 The Significance of a QoS Aware MPEG-4 Video Delivery Algorithm in the Context of Video over WLAN

As discussed in chapter 2, there are different video standards (MPEG-1, MPEG-2, MPEG-4, H.263, H.264, UHD etc.) available which have different QoD requirements (in terms of bandwidth, delay, loss rate etc.) when streamed over WLANs. Video can be broadly categorized into real time video (interactive video, e.g. video conferencing) and streamed video which have different loss and delay requirements. This work is specifically concerned with streamed MPEG-4 video over IEEE 802.11b/e networks. Latency is a relatively less important issue for streamed video traffic compared to real time video. If there are uncontrolled packet losses in the medium then pixilation or loss of video frames or loss of audio/video synchronization severely reduces the QoS experienced by the clients.

It has been stated in earlier chapters that IEEE 802.11b WLANs are unable to service many bursty video streams simultaneously with acceptable levels of QoS. Video applications are increasingly becoming more demanding of network resources, e.g. the progression from SD to HD to UHD videos requires more and more network bandwidth. QoS enabled IEEE 802.11e/WMM or the relatively recent introduction of IEEE 802.11n networks certainly improve streamed video performance to some extent, but are not sufficient to guarantee reliable video delivery under dynamic and heavily loaded shared wireless conditions. Hence, real time and streamed video traffic requires a different treatment from other traffic on the WLAN. So it becomes important to manage the transmission of the video traffic on a WLAN carefully to achieve acceptable levels of QoS within the system constraints.

Therefore an effective QoS aware MPEG-4 video delivery algorithm must allow a network engineer to address the degradation of streamed video quality over WLANs. When there are insufficient resources available on the medium for multiple users, the packet loss rate, delay, jitter etc. might exceed the acceptable levels as specified by ITUT [36,37]. In the absence of any control mechanism the video quality can potentially suffer from sudden and catastrophic drops in quality when streamed over WLANs. An efficient QoS aware MPEG-4 video delivery algorithm should eliminate unpredictability and provide the most favourable operating conditions for the video streams under the prevailing network conditions.

4.2 Video Structure and WLAN

Detailed technical discussions regarding the WLANs and different video encoding configurations were presented in chapter 2. In this section the nature of the interaction between video traffic and WLAN performance will be further discussed.

4.2.1 *IPB* Frames Hierarchy

The *IPB* frame based nature of MPEG-4 video, the relative priorities between the frames and related concepts were discussed in detail in chapter 2. Video characteristics have a huge impact on the performance of WLANs. According to the MPEG-4 standard, a frame is generated every 40 ms, i.e. the frame rate of MPEG-4 videos is 25 frames per second. It has been explained that in this video encoding standard the ideal ratio of *I, P, B* frames is *I*:*P*:*B* = 1:4:10 in a 15 frame long GOP (PAL system). A GOP begins with an *I* frame followed by two *B* frames and then a *P* frame is followed by two *B* frames. The pattern *PBB* repeats afterwards resulting in ***IBB PBB PBB PBB PBB*** pattern for a typical GOP as shown in Fig. 4.1. Hence a GOP can be described as being composed of five frames triplets (the first triplet is *IBB*, the four other triplets have a pattern *PBB*).

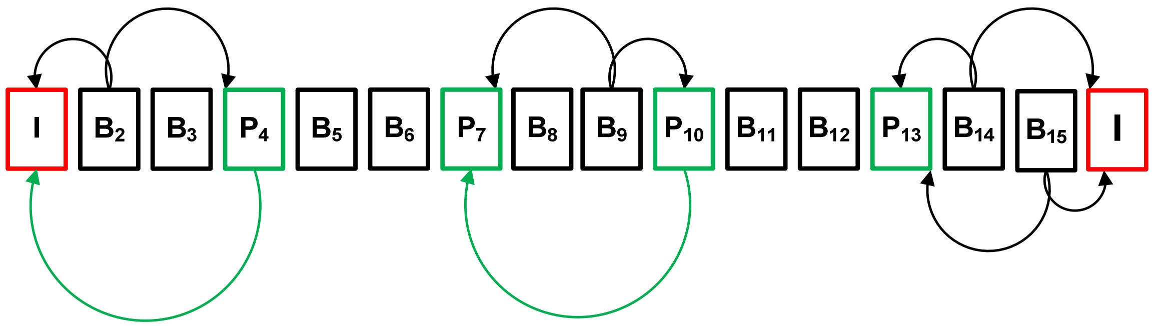

Fig. 4.1: Interdependency of The MPEG-4 Video Frames within A GOP

According to the MPEG-4 standard the most important frame is the *I* frame, then the relative priority goes on to *P* and *B* frames ($I > P > B$). An *I* frame can be decoded independently, a *P* frame requires the previous *I* or *P* frame to be decoded. *B* frames require the presence of the previous and next *I* or *P* frames. As shown in Fig. 4.1, B_2 and B_3 frames require the presence of *I* and P_4 frames, P_4 and P_7 frames need respectively *I* and P_7 frames, B_5 and B_6 frames require both P_4 and P_7 to be decoded correctly. Hence It can be concluded that the earlier frames have more importance than the later frames within a GOP of the same category ($P_4 > P_7 > P_{10} > P_{13}$). From a GOP triplet point of view the earlier triplets are more important than the later ones. If the *I* frame is lost then transmitting the remaining fourteen frames of that GOP represents a waste of bandwidth as they would not be decodable at the client. Similar arguments could be made for the *P* frames, e.g. if a P_4 frame is lost then the remaining P_7 , P_{10} , P_{13} and B_5 - B_{15} frames would become undecodable resulting in a waste of valuable WLAN resources. This frame interdependency will be exploited later in the development of the proposed QoS aware MPEG-4 video delivery algorithm.

4.3 The Proposed QoS Aware MPEG-4 Video Delivery Algorithm for Streamed Video

The proposed QoS aware MPEG-4 video delivery algorithm described in this section will be primarily dealing with the performance aspect related to frame losses. As described in detail in chapter 2, there are three ways in which video frames can get lost on a WLAN. These are – MAC collisions, buffer overflow and transmission errors. Frame loss due to MAC collisions and MAC buffer overflow has only been considered in this work. Transmission losses were not investigated, i.e. a lossless channel was assumed. Transmission errors arising from MAC collisions can be reduced significantly by effectively retransmitting the failed frames at the expense of a reduced buffer service rate and an increased bandwidth requirement. On the other hand, buffer overflow losses i.e. once a frame is lost from the buffer due to a lack of buffer space, cannot be recovered. From the point of view of QoS, buffer overflow (BO) is potentially catastrophic and should be avoided at all cost. In this respect a QoS aware MPEG-4 video delivery algorithm is introduced in this section with the objective to eliminate (in so far as it is possible) buffer overflow and to minimise frame retransmissions to save bandwidth resources. It employs two mechanisms namely frame retransmission (ReTx) and GOP truncation (GOPT) sequentially. The algorithm aims to replace uncontrolled packet loss due to MAC collisions and buffer overflow by a controlled packet loss scheme that permits a graceful degradation in MPEG-4 video quality when streamed over IEEE 802.11b networks. Thus the proposed scheme implements an *Adaptive Video Streaming Scheme* by determining the best way to combine the

ReTx and GOPT mechanisms. The proposed novel algorithm is also generic in nature, i.e. as it is concerned with buffer occupancy; it would work with all types of IEEE 802.11 based WLANs (e.g. b/g/a/n). The process sequence of the proposed QoS aware MPEG-4 video delivery algorithm involves firstly applying ReTx to address the loss rate (i.e. to achieve a loss rate of $<\leq 1\%$) and then to apply GOPT to address the buffer occupancy (i.e. to achieve an average zero buffer occupancy) as illustrated in Fig. 4.2.

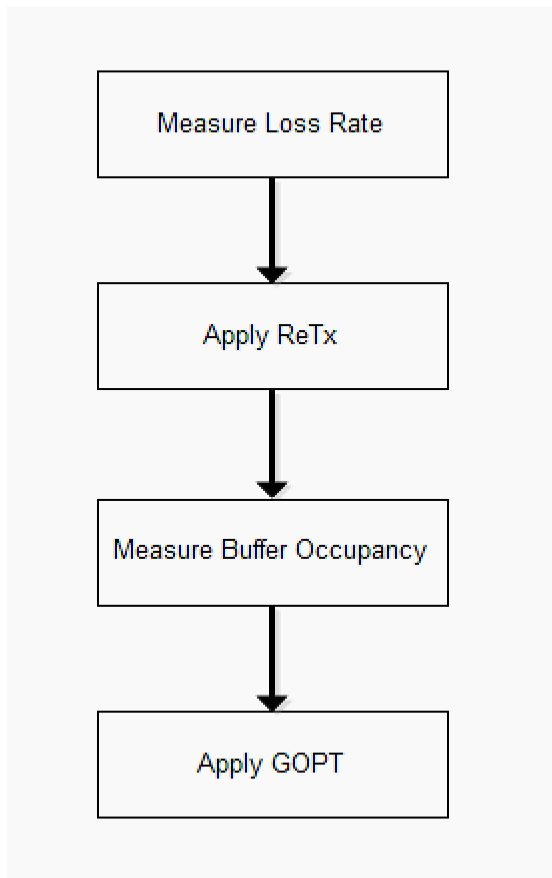

Fig. 4.2: The Sequence Diagram of the Proposed QoS Aware MPEG-4 Video Delivery Algorithm

4.3.1 The ReTx Mechanism

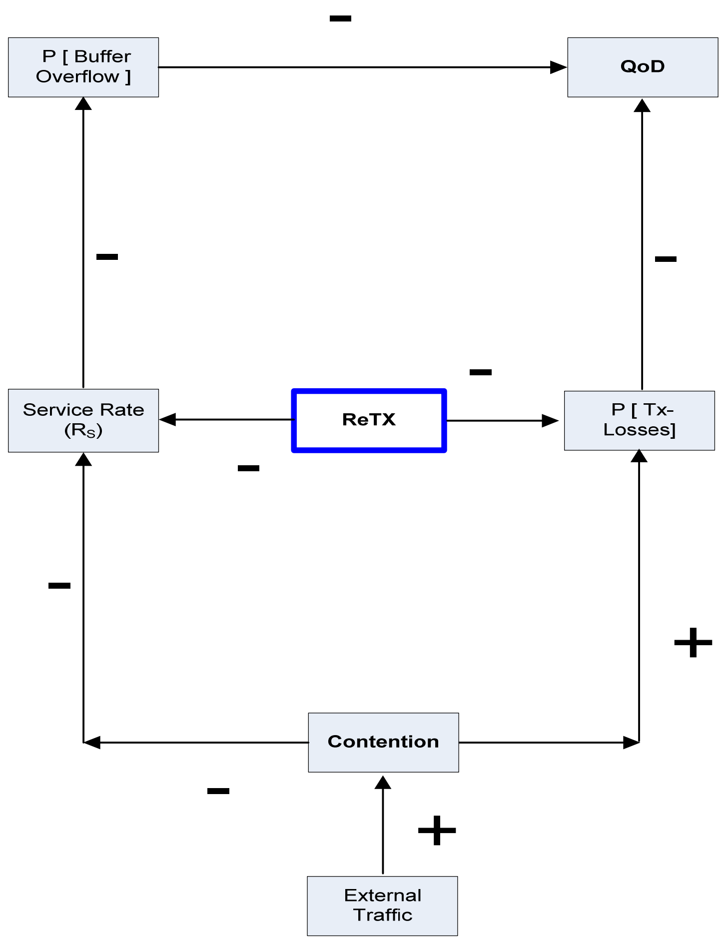

Fig. 4.3: Modelling The ReTx Operation

In a WLAN, if no ACK is received the sender will retry to transmit (using the normal CSMA/CA procedures) until either successful or the operation is abandoned with exhausted retries. MAC frame retransmission is an effective technique as well documented in the literature as discussed in chapter 3. In summary it trades off bandwidth cost for greater video quality, i.e. it's a mechanism for enhancing reliability and hence QoS at the price of increased BW requirement. When frame retransmission is applied, it consumes more bandwidth but provides for higher quality video.

Fig. 4.3 describes the operation of the ReTx. As depicted using block diagrams, the probability of MAC collisions increases with increased contention in the system which is related to the relative packet rates of the competing stations. When the level of contention is increased the buffer service rate decreases and the probability of buffer overflow increases. As a result probability of packet loss increases resulting in reduced video QoD. The ReTx mechanism facilitates the management of frames losses due to contention. Consequently the QoD of the video is increased due to reduced probability of transmission losses at the expense of higher probability of buffer overflow by reducing the effective average buffer service rate. This effectively increases the QoD (as corrupted frames are retransmitted) by ensuring that frames are successfully received thereby minimising the transmission losses and achieving controlled frame losses. But at the same time the buffer service rate is further reduced and hence the probability of buffer overflow is increased. The GOPT mechanism, described in the next section reduces the probability of buffer overflow at the expense of a reduced quality by discarding the comparatively less important video frames in triplets (*PBB*). A ReTx scheme has been implemented to achieve the acceptable 1% frame loss rate for streamed video. The target frame loss rate of 1% is defined by the ITUT [36, 37]. In this work a target of achieving < 1% loss rate for *I, P,* and *B* frames individually was adopted.

4.3.2 The GOPT Mechanism

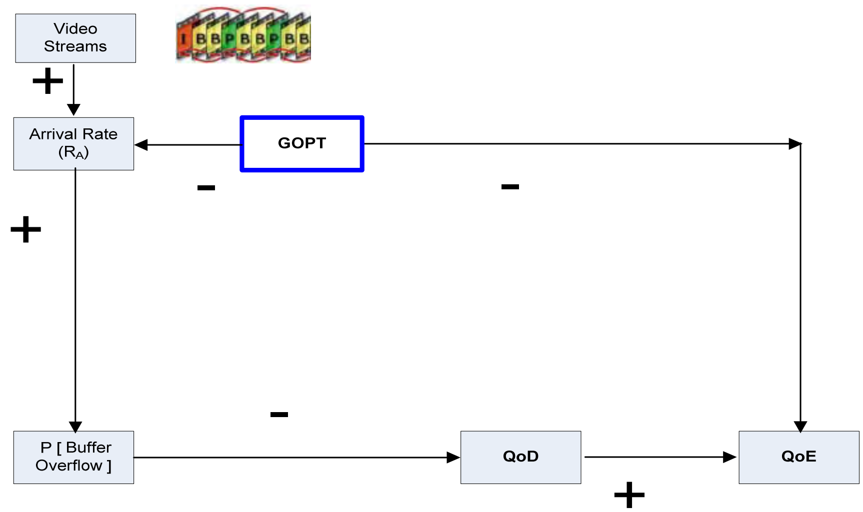

Fig. 4.4: Modelling The GOPT Operation

As mentioned in earlier sections that the proposed QoS aware MPEG-4 video delivery algorithm employs two mechanisms namely frame retransmission (ReTx) and GOP truncation (GOPT). It has been detailed in the earlier section that when the ReTx mechanism is employed it increases the probability of buffer overflow as ReTx requires extra bandwidth. After implementing frame retransmission, the second mechanism, namely GOPT is employed. It is novel and unique and involves in reducing the probability of buffer overflow. It works by selectively dropping frame triplets in order of their importance from the GOP to reduce the number of video packets required to be transmitted. The GOPT trades off quality for bandwidth. The QoS aware MPEG-4 video delivery algorithm proposes that if not all frames can be serviced in a timely manner, a QoS delivery strategy might be employed where the higher priority frames would be transmitted and the lower priority ones would be discarded, i.e. transmitting the more important GOP triplets instead of the whole GOP (described in the earlier section). Fig. 4.4 describes the methodology of the GOPT mechanism. As evident from the figure, when the GOPT is employed, it artificially reduces the arrival rate of incoming frames (Hence, the probability of buffer overflow is decreased.) as a smaller number of frames need to be transmitted to counter the increased probability of buffer overflow when ReTx is used. But the price of employing GOPT is that this mechanism improves the QoD of the transmitted video at the expense of QoE. Discarded GOP triplet frames must be employed in such a way that the truncation scheme has the minimum effect on the received video stream's quality at the client.

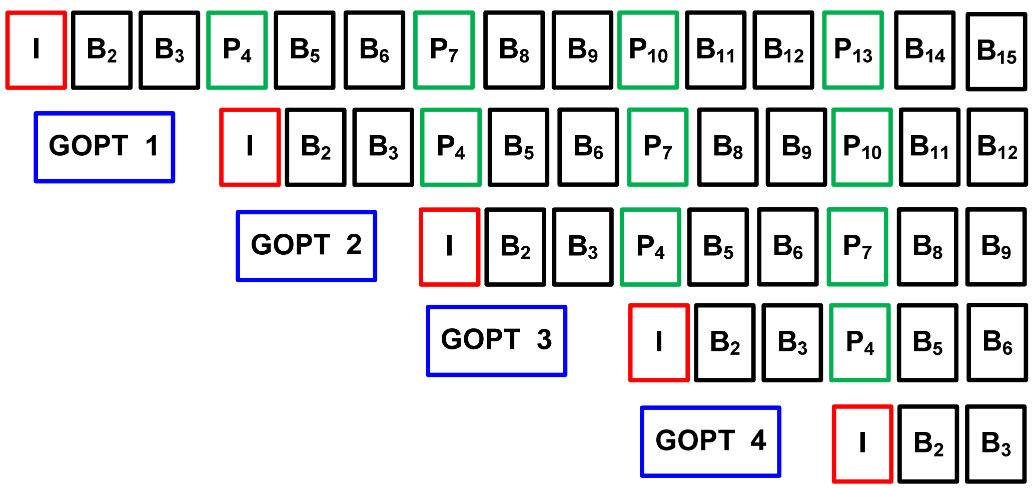

Fig. 4.5: Description of GOP Truncation (GOPT)

The QoS aware MPEG-4 video delivery algorithm proposes that to avoid buffer overflow the last GOP triplet ($P_{13}B_{14}B_{15}$) which is the least important of the five triplets is discarded first. If this is not sufficient to eliminate the probability of overflow then the second last triplet ($P_{10}B_{11}B_{12}$) is discarded and so on until the probability of buffer overflow is eliminated. This has been shown in Fig. 4.5. In summary GOPT is the mechanism to reduce the probability of buffer overflow by reducing the number of packets to be transmitted after employing the ReTx mechanism. It allows the system to achieve a graceful degradation in video quality. Error concealment within the GOP has not been considered in this work.

4.3.3 The Inter-relationship of the ReTx and GOPT Mechanisms

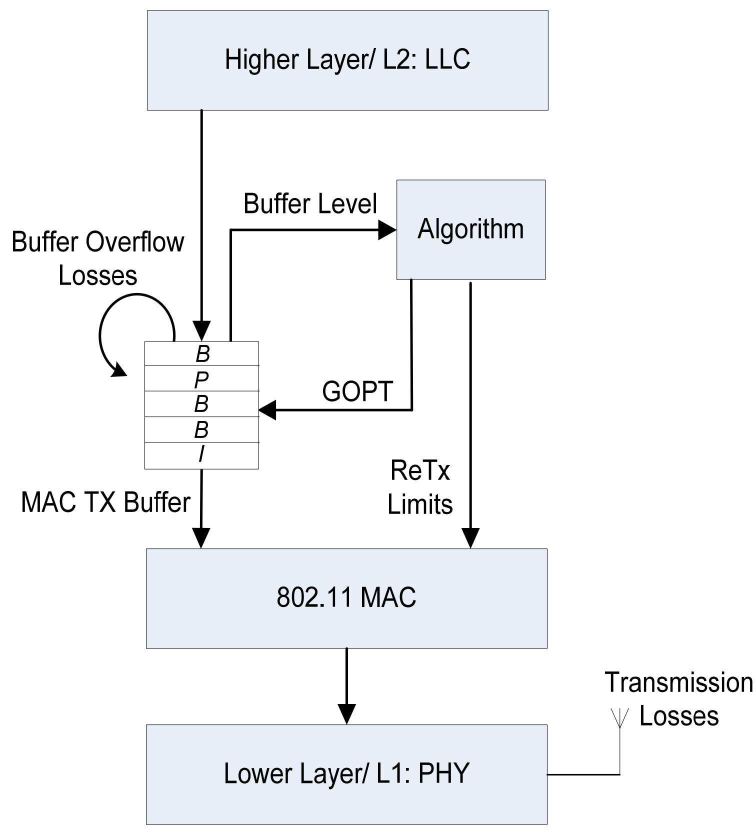

Fig. 4.6: ReTx and GOPT Mechanisms of The QoS Aware MPEG-4 Video Delivery Algorithm

As described in chapter 2, frames get lost in the WLAN medium in uncontrolled ways resulting in degradation in video quality. Fig. 4.6 shows both the mechanisms of the QoS aware MPEG-4 video delivery algorithm to address the losses. As depicted in the figure, the algorithm works by combining the GOP truncation (GOPT) and frame retransmission (ReTx) to minimise the probability of uncontrolled frame loss of the video streams at the expense of a reduced quality of experience (QoE). It operates with the goal of eliminating the probability of buffer overflow and the loss rate in the system. In WLANs frames loss occurs due to MAC collisions, buffer overflow etc. thereby reducing the streamed video QoD. At first, to reduce the loss rate to an acceptable level ($\leq$1%), frame ReTx mechanism is employed. This enhances the quality of the streamed video at the expense of higher probability of buffer overflow. Then to reduce the probability of buffer overflow, the GOPT mechanism is utilised. The QoS aware MPEG-4 video delivery algorithm gets information about the level of buffer occupancy and loss rate as inputs and then decides the level of frame ReTx and GOPT to eliminate buffer overflow.

The average buffer frame arrival rate is decreased by the GOPT mechanism thereby reducing the probability of buffer overflow at the expense of reduced QoE by not transmitting the least important frames within a GOP. Hence the algorithm aims to achieve a trade off between these two mechanisms (GOPT and ReTx) in order to eliminate buffer overflow and minimize transmission losses. This ensures the realisation of the most favourable network conditions for the delivery of MPEG-4 video frames on WLANs.

Table 4.1 summarises the various mechanisms and their interactions in the proposed QoS aware MPEG-4 video delivery algorithm as discussed in detail in the above sections. Fig. 4.7 presents a complete architecture of the proposed algorithm by combining Fig. 4.3 and 4.4. Chapter 5 will describe the implementation and validation of the proposed algorithm.

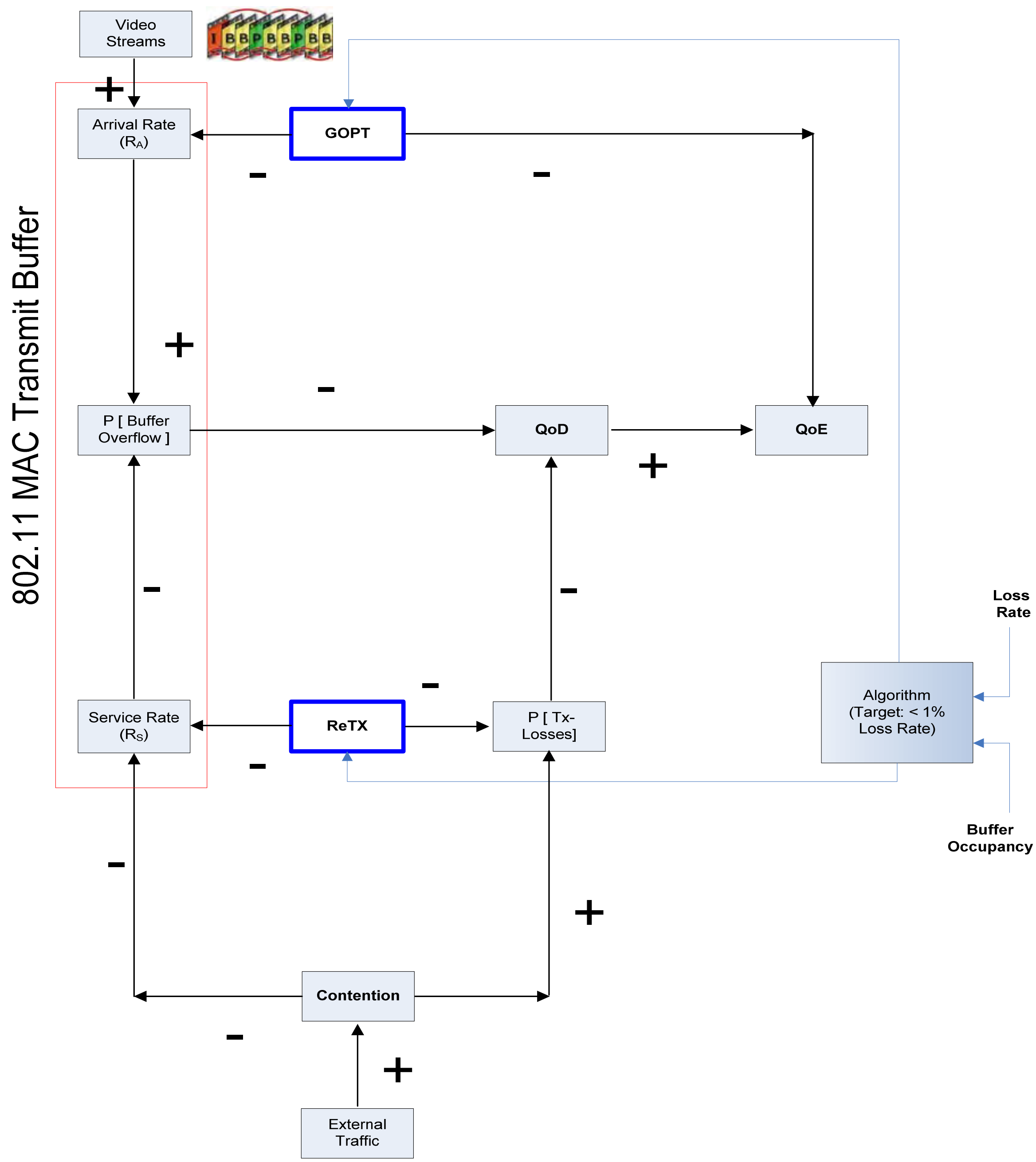

Fig. 4.7: An Architecture of The Proposed QoS Aware MPEG-4 Video Delivery Algorithm (Showing The ReTx and GOPT Trade Offs)

Table 4.1: Summary of the Various Mechanisms and Model Parameters of the Proposed QoS Aware MPEG-4 Video Delivery Algorithm.

Model Parameters / Mechanisms	Arrival Rate (R_A)	Service Rate (R_S)	*P* [Buffer Overflow]	*P* [Trans-mission Losses]	QoD (Quality of Delivery)	QoE (Quality of Experience)	Trade-off Involved	Comments
External Contention	-	Increased contention reduces the service rate.	Causes the probability of buffer overflow to increase due to reduced service rate.	Due to increased collisions probability of transmission losses increases.	Due to increased collisions QoD is decreased.	Reduced due to reduced QoD.	-	Contention depends on the relative packet rates of the competing stations.
Frame Re-transmission	-	Reduces the effective service rate due to delay in re-transmitting packets.	As service rate is reduced, probability of buffer overflow is increased.	Probability of transmission losses is reduced as corrupted frames are retransmitted	Increased due to reduced probability of transmission losses.	Due to increased QoD, QoE increases.	Retransmission allows one to trade off BW cost against the QoS benefit.	Mechanism for enhancing reliability and hence QoS at the price of increased BW requirement.
GOP Truncation	Reduces the effective arrival rate.	-	Reduced probability of buffer overflow due to reduced arrival rate.	-	-	QoE is decreased due to removal of frames from the GOPs (Graceful Degradation).	Trade off against QoE to avoid BO.	Mechanism to reduce the probability of buffer overflow by reducing the number of packets to be transmitted.

Chapter 5 Validation of the Proposed Novel QoS-Aware MPEG-4 Video Delivery Algorithm to Improve the Streamed Video QoD over the Lossy IEEE 802.11b WLANs by Exploiting The *IPB* Frame Based Nature of the MPEG-4 Videos

5.1 Implementation of the QoS Aware MPEG-4 Video Delivery Algorithm in *C* Programming Language

To validate the QoS aware MPEG-4 video delivery algorithm two network scenarios will be described in later sections, namely uplink and downlink network topologies in the presence of 1500 byte size CBR background traffic. Separate programs have been written in *C* programming language for Uplink and Downlink IEEE 802.11b networks to simulate MPEG-4 video streamed over WLANs. Developing programs was preferred to running simulations in various publically available simulation software (e.g. NS-3/OPNET/OMNET etc.) as it provides for greater flexibility and convenience. The programs will be discussed in detail in later sections. A test suite of 12 real video clips (selected from 6 different genres) with varying degrees of spatial and temporal complexities were chosen to extract modelling parameters which would be used in the traffic generator as part of the simulation process. The output of the simulation has been analysed by programs written in *Perl* and *C* programming languages.

The main building blocks in implementing the QoS aware MPEG-4 video delivery algorithm are –

1. Detailed Analysis of the Video Clips for Extracting Modelling Parameters
2. Modelling Incoming Video and Background Traffic.
3. Developing a MAC Model.
4. Data Collection after the MAC operation and the QoS aware MPEG-4 video delivery algorithm implementation for Uplink and Downlink Networks to Evaluate Performance of the Streamed Video.

For the downlink scenario the performance of the QoS aware MPEG-4 video delivery algorithm will be illustrated by demonstrating considerable bandwidth saving while achieving the target <1% loss rate. In the uplink scenario the performance is indicated by the loss rate to investigate the QoD by setting a target of zero buffer occupancies at the video queues and demonstrating that various levels of GOPT would be required for different video contents to achieve that target.

5.1.1 Detailed Analysis of the Video Clips for Extracting Modelling Parameters

Experimental results with different types of real video clips over real WLANs have been presented in chapter 4. However to broaden the range of test scenarios twelve new video clips from six different genres have been used for validating the QoS aware MPEG-4 video delivery algorithm as they represent a range of content complexity. All these movie clips have been chosen in such a way so that they exhibit different characteristics, e.g. scene change frequency, scenes with varying light levels, motion, strong colours, hard edges etc. Various modelling parameters have been extracted from these video clips for using in the computer simulation. These clips (duration 5 minutes each) were collected from different sources which are listed below –

a) CGI/ Sci-Fi : AVATAR, 2012.
b) Action : DIE HARD 4, KING ARTHUR.
c) Animation : LION KING, ICE AGE 2.
d) Sport : RUGBY (courtesy: RTE, Ireland), FOOTBALL (courtesy: FIFA, Brazil vs. North Korea Game, World Cup 2010).
e) Documentary: BBC PLANET EARTH: ICE WORLDS, THE ANTARTICA CHALLENGE.
f) Talking Head: Interviews of Hollywood actor MATT DAMON (courtesy: CBS News, USA) and Facebook founder MARK ZUCKERBERG (courtesy: Stanford University, USA).

The software application FFMPEG [152] has been used to convert all the collected video clips into MPEG-4 format videos. After conversion all the clips had the following target characteristics –

Video:

Codec	: mp4v
Display Resolution	: 720 x 576 pixels (PAL), Advanced Simple Profile (ASP)
Duration	: 5 minutes (300 sec)
Frame Rate	: 25 fps
GOP Size	: 15
GOP pattern	: *IBB PBB PBB PBB PBB (where I:P:B* = 1:4:10)
B frame freq	: 2

Audio:

Sample Rate	: 44100 Hz 16 bits Stereo
Bit Rate	: 64 kbits/sec
Codec	: mp4a

These target parameters were chosen as they are typical for MPEG-4 video applications. The characteristics of AVATAR clip is described in the next section as this particular clip has the largest frame size of all the clips.

5.1.1.1 AVATAR Movie Clip Analysis

Fig. 5.1: An AVATAR Movie Snapshot

After analyzing the ‘AVATAR’ clip (Fig 5.1) using computer programs written in *Perl* language, it was found out that the 300 second clip contains 567 *I* frames, 1832 *P* frames, and 4795 *B* frames. The average sizes of the *I, P,* and *B* frames have been calculated at 9952, 6159, and 3832 bytes respectively.

As discussed in chapter 2, according to the IEEE 802.11b standard, a station intending to transmit a packet senses the medium to find out if the medium is idle through a period of time called *DIFS* (50 μs). The minimum and maximum values of the CW are 32 and 1024 respectively. If the transmission is successful, the receiving station waits for a *SIFS* time duration (10 μs) and sends an ACK frame. The process is complete when the sending station receives the ACK successfully.

The Logical Link Control (LLC) sub-layer is the upper portion of the data link layer of the OSI Model and presents a uniform interface to the user of the network layer. IEEE 802.11 relies on logical-link control (LLC) encapsulation to carry higher level protocols. Beneath the LLC sub-layer is the Media Access Control (MAC) sub-layer, which is dependent on the particular medium being used (Ethernet, token ring, 802.11, etc.). Hence in addition to the payload data, there are additional bytes of data added in the encapsulation process. The 802.11 MAC header adds data for various control and management functions, error detection, and addressing. Further bytes are added by the LLC/SNAP (Sub-network Access Protocol) encapsulation header to identify the network layer protocol. To multiplex higher-level protocol data over the wireless link, IEEE 802.11 uses the LLC/SNAP encapsulation. SNAP headers begin with a *destination service access point* (DSAP) and a *source service access point* (SSAP). After the addresses, SNAP includes a Control header. The last field inserted by SNAP is an organizationally unique identifier (OUI).

The whole process is summarised in Fig. 5.2 -

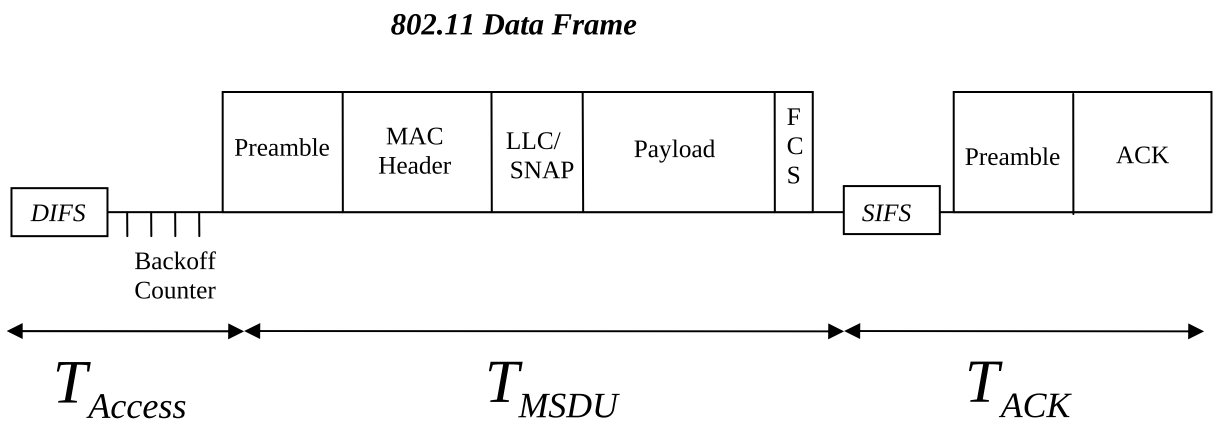

Fig. 5.2: An IEEE 802.11 Frame Transmission under the CSMA/CA Process

Total time required to transmit a data packet is-

$$T_{Frame} = T_{Access} + T_{MSDU} + T_{ACK} \quad (5.1)$$

Frame transmission times (T_{MSDU}) for different types of *I, P, B* frames have been calculated according to the following equation –

$$T_{MSDU} = Preamble + \frac{MACHeader + LLC/SNAP + Payload + FCS}{LineRate}$$

$$T_{MSDU} = 192 or 96 \mu s + \frac{(24+8+20+8+L_p+4)x8}{LineRate}$$

$$T_{MSDU} = 192 or 96 \mu s + \frac{512 + 8xL_P}{LineRate} \quad (5.2)$$

[All frame sizes in bytes, 192 μs for Long and 96 μs for Short Preamble, Line Rate in Mbps, L_p = Payload, Various associated overheads: MAC header 24 bytes, UDP header 8 bytes, IP header 20 bytes, LLC/SNAP header 8 bytes, and FCS (Frame Check Sequence) 4 bytes. The FCS allows stations to check the integrity of received frames respectively.]

$$T_I = 96\mu s + \frac{512 + 8x9952}{11} = 7381\mu s$$
$$T_P = 96\mu s + \frac{512 + 8x6159}{11} = 4622\mu s$$
$$T_B = 96\mu s + \frac{512 + 8x3832}{11} = 2930\mu s$$
$$T_{ACK} = SIFS + Pr\ eamble + ACK = (10 + 96 + \frac{14x8}{1})\mu s = 208\mu s$$

[In IEEE 802.11b, line rates for Data and ACK frames are 11 and 1 Mbps respectively.]

Appendix-A contains detail analysis of the remaining eleven video clips. The analyses for all the clips are summarised in tables 5.1(a) – 5.1(c). The tables contain number of different type of frames, average frame sizes and, total transmission time required for all frames types for 12 video clips. Theoretically a typical MPEG-4 video clip of 300 second duration would contain 500 *I* frames, 2000 *P* frames, and 5000 *B* frames. But it was noted that real life video clips do not always follow this practice always. The number of *I* frames varied from around 500 to 600 range. The numbers of *P* and *B* frames follow this pattern in proportionate manners. It can be seen that the CGI/Action movies have larger average frame sizes while the talking head video clips have smaller average frame sizes. The average frames sizes of the frames are larger than typical MTU of the Ethernet (1500 bytes). Hence in reality they would be fragmented while being transmitted over networks, however fragmentation was not considered in this work. It would take longer to transmit larger video frames over the IEEE 802.11 networks.

Table 5.1(a): Various Frame Counts for All 12 Video Clips

Frame Count (#)		CGI		Action		Animation		Sport		Documentary		Talking Head		ALL (Avg.)
%		AVA	2012	DH	KA	LK	IA	RUG	FB	BBC	ANT	MD	MZ	
I	#	567	580	589	596	538	535	520	507	540	528	498	502	542
	%	7.88	7.73	7.85	7.94	7.17	7.13	6.93	6.77	7.20	7.04	6.68	6.69	7.25
P	#	1832	1921	1912	1907	1964	1967	1982	1992	1962	1974	1987	2002	1950
	%	25.47	25.61	25.49	25.41	26.18	26.22	26.42	26.59	26.15	26.31	26.67	26.67	26.10
B	#	4795	5000	5000	5001	4999	5000	5000	4992	5000	5000	4966	5003	4980
	%	66.65	66.66	66.66	66.64	66.65	66.65	66.65	66.64	66.65	66.65	66.65	66.64	66.65
Total	#	7194	7501	7501	7504	7501	7502	7502	7491	7502	7502	7451	7507	7472
	%	100	100	100	100	100	100	100	100	100	100	100	100	100

Table 5.1(b): Average Frame Sizes for All 12 Video Clips

Average Frame Size (byte)	CGI		Action		Animation		Sport		Documentary		Talking Head		ALL (Avg.)
	AVA	2012	DH	KA	LK	IA	RUG	FB	BBC	ANT	MD	MZ	
***I*_Avg**	9952	7310	7213	7427	6593	6861	6607	7994	7605	7904	7657	7909	7586
***P*_Avg**	6159	4167	3728	5158	2726	2489	3603	2550	2566	1704	1307	1388	3129
***B*_Avg**	3832	2420	2318	3325	1364	1374	1684	1190	1381	849	735	666	1762
(*I*_Avg + *P*_Avg + *B*_Avg) /3	6648	4633	4420	5304	3561	3575	3965	3951	3851	3486	3233	3321	4159

Table 5.1(c): Total Transmission Delay Calculations for The *IPB* Frames of All 12 Video Clips

T_{MSDU} μs	CGI		Action		Animation		Sport		Documentary		Talking Head		ALL (Avg.)
	AVA	2012	DH	KA	LK	IA	RUG	FB	BBC	ANT	MD	MZ	
I	7381	5459	5389	5544	4938	5133	4948	5957	5674	5891	5711	5895	5660
P	4622	3173	2854	3894	2125	1953	2763	1997	2009	1382	1093	1152	2418
B	2930	1903	1828	2561	1135	1142	1367	1008	1147	760	677	627	1424

5.1.2 Modelling Incoming Video and Background Traffic

Incoming video and background traffics have been modelled in the simulation. Streamed video/background stations start at random times. The time difference between any two successive video frames for a particular station is 40 ms to reflect the MPEG-4 video standard. There are three types of video frames, namely *I, P, B* which have different frame sizes and frame transmission times as calculated from the real video clips (details in section 5.2). The GOP length is 15 with a frame pattern - ***IBB PBB PBB PBB PBB***. Each video frame has a unique frame number, GOP number, buffer position and time scale associated with it. The simulation is run for 300 seconds. Different load levels of video and CBR background (of packet size 1500 bytes) traffics have been implemented.

5.1.2.1 Downlink Scenario

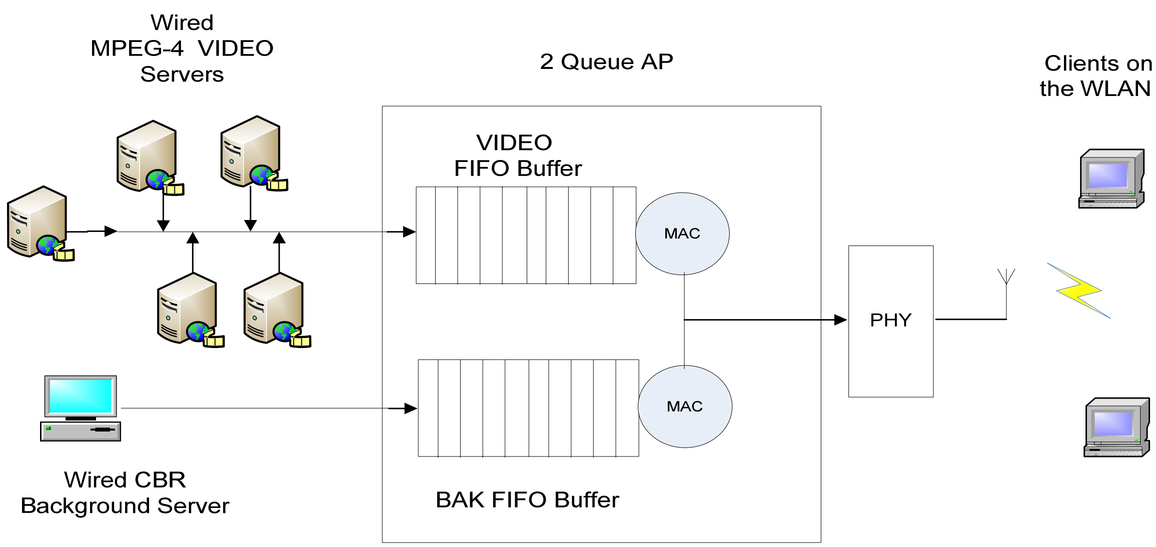

Fig. 5.3: Downlink Configuration

A two queue system has been implemented (Fig 5.3) for analysing the downlink scenario. The video and background frames generated from all the video stations and background station respectively are sent to the video and background queue. The MPEG-4 Video Servers and the Background Traffic Generator are started at different random times. The two MAC queues within the same AP contend with each other to access the medium and transmit packets. The queue that has the smaller back-off counter value wins the transmission opportunity and transmits the packet present at the front of its queue. For both uplink and downlink scenarios, each and every video and background frame from all the different stations is tracked from its source until its transmission from the MAC transmit queues for later analysis and implementation of the proposed QoS aware MPEG-4 video delivery algorithm.

5.1.2.2 Uplink Scenario

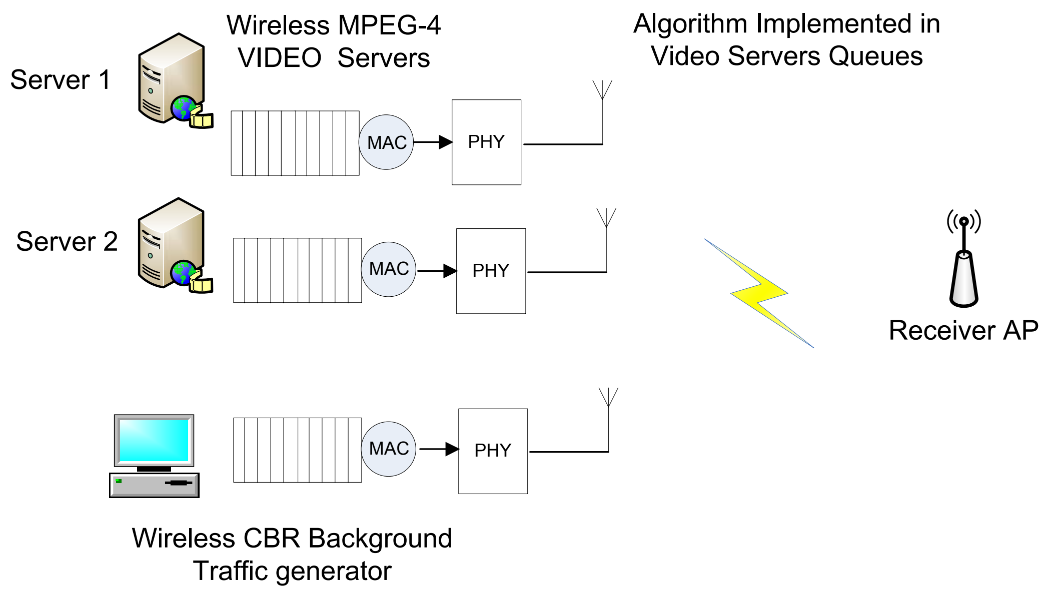

Fig. 5.4: Uplink Configuration

Frames are generated and pushed into the back of the corresponding queues to implement FIFO MAC buffers which contend with each other in accessing the medium through the CSMA/CA MAC protocol. Each video/background station has a single queue and transmits one video stream (Fig 5.4). So if there are five stations contending to access the medium, there are five MAC buffer queues and five individual video/background traffic generators. The PAL system defines a GOP size of 15 frames where there are 1 *I* frame, 4 *P* frames, and 10 *B* frames. As a PAL system (25 fps) has been simulated, 500 GOPs (500x15 i.e. 7500 video frames: 500 *I*, 2000 *P* and 5000 *B* frames) per station were generated over 300 seconds.

5.1.3 Building an IEEE 802.11b MAC Model

The IEEE 802.11b MAC mechanism has been implemented. When there is a frame present in a station's MAC buffer, the station senses the medium to establish if it is busy or idle. If the medium is found to be idle then stations wait for a time known as *DIFS* and generate BC values. The BC is initialised by randomly choosing an integer from a contention window (CW). The decrementing of the BC is frozen when the station senses the medium is busy and is resumed when the medium is free for a time period of a *DIFS*. When a station's BC reaches zero, it transmits its packet. When several stations are attempting to transmit, the station that picks the lowest random number wins. If two or more stations transmit at the same time, a collision occurs. The collision is resolved by having the stations involved restarting their random access processes again, but with a CW that has been doubled. Contention window sizes are always 1 less than an integer power of 2 (e.g., 31, 63, 127, 255, 511, and 1023). The contention window is reset to its minimum size when a frame is transmitted successfully, or the associated retry counter is reached, and the frame is discarded. When the packet reaches the destination, the destination station waits for a time *SIFS* and then it sends an acknowledgment (ACK) frame to the sending station to announce that the transmission was successful. When the medium is busy, all other stations must wait for the channel to become idle. All stations maintain a random back-off interval counter which they start decrementing when the medium is sensed idle, i.e. after the transmission has finished. When the transmitting station receives an ACK after transmitting the frame, all stations start decrementing their back-off counters again after waiting for a time of *DIFS*. The flowchart in Fig. 5.5 shows the implementation of IEEE 802.11b [153].

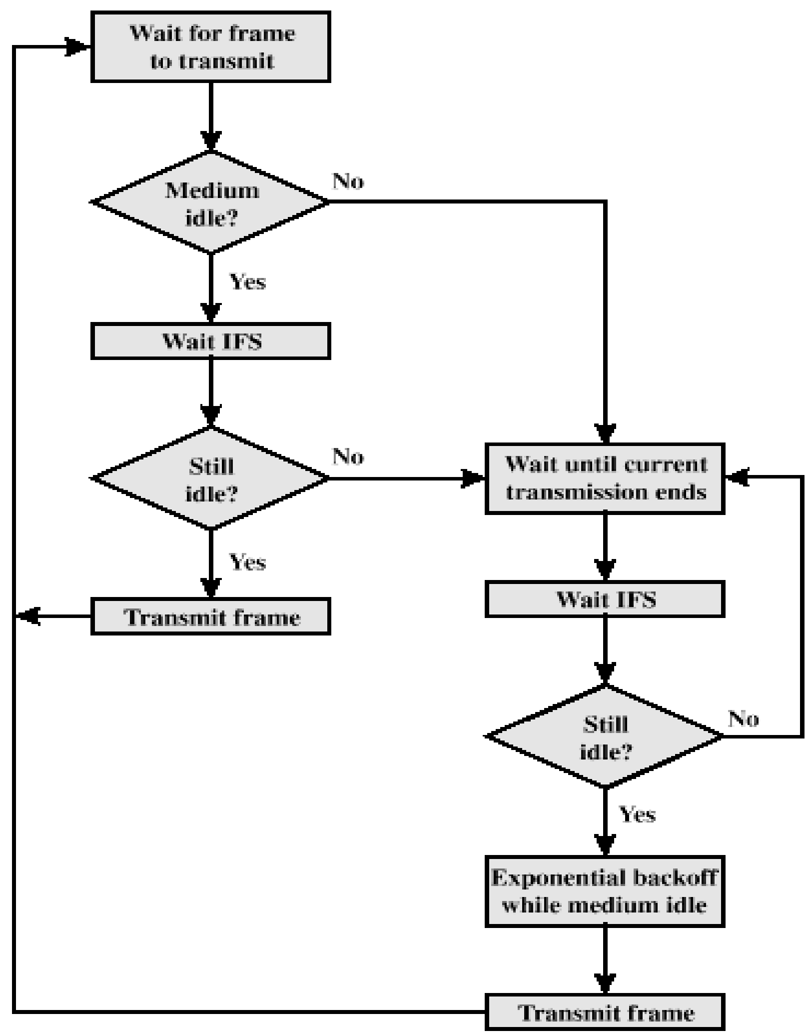

Fig. 5.5: The Flowchart Shows the Implementation of IEEE 802.11b

5.1.4 Data Collection and the Implementation of the QoS Aware MPEG-4 Video Delivery Algorithm for Uplink and Downlink Networks to Evaluate Performance of the Streamed Video.

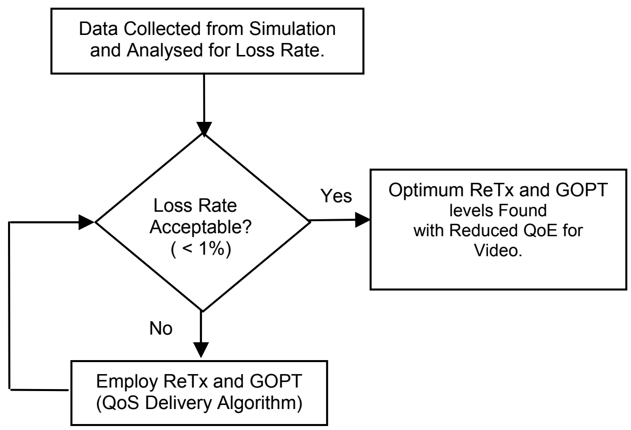

Fig. 5.6: Proposed QoS Aware MPEG-4 Video Delivery Algorithm Implementation

Computer programs written separately in *Perl* and *C* languages have been used for analysis of the simulated data. Before executing the MAC simulator, various metrics (e.g. pdf, frequency, size etc.) regarding the GOPs and various types of frames has been analysed for all 12 video clips. Based on the MAC operation, after each attempt it is necessary to identify the frame type (*I, P, B*) that has been either transmitted or collided. Hence, the *C* program pops corresponding video/background frames from the front of the related MAC buffer queue(s) after each transmission attempt. Various counters have been realised in the program to record detailed information of all successful and unsuccessful (i.e. collided) frames associated with time stamps, frame sequence, frame type, GOP number etc. for all stations and for all attempts. ReTx and GOPT mechanisms have been realised in the *C* program. The fundamentals of the implementation described above is shown in Fig. 5.6.

5.1.5 Block Level Diagram of the Implementation Details

The following block diagram (Fig. 5.7) summaries the implementation of the simulator that has been discussed in detail in earlier sections -

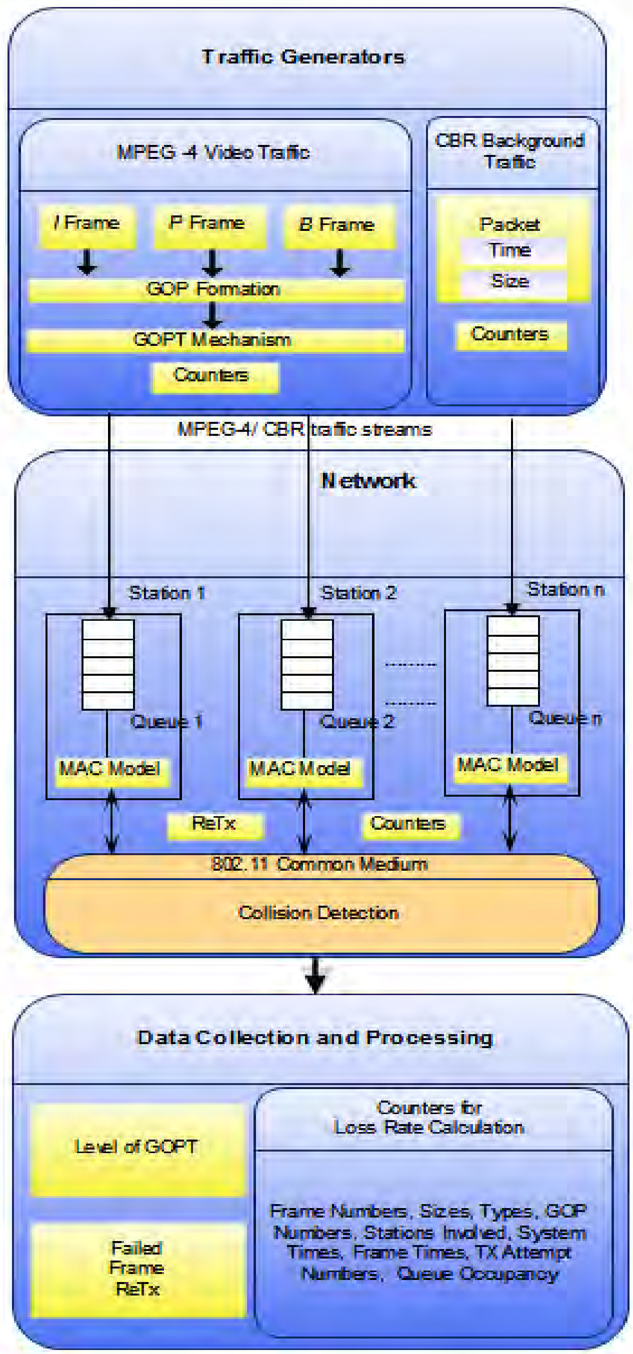

Fig. 5.7: Block Diagram of the Proposed QoS Aware MPEG-4 Video Delivery Algorithm Implementation

5.1.6 Validation of the MAC Simulator

As described in earlier sections the proposed QoS aware MPEG-4 video delivery algorithm has been implemented in the *C* programming language. Furthermore, an IEEE 802.11b MAC simulator has been realised for this purpose. The MAC simulator has been benchmarked against an established simulator known as NS-3 (Network Simulator Version 3) [154]. Experimental scenarios were designed and evaluated on NS-3 and the MAC simulator developed for this thesis. This benchmarking exercise shows that the simulated results were in good agreement for both cases.

5.1.6.1 Experimental Setup

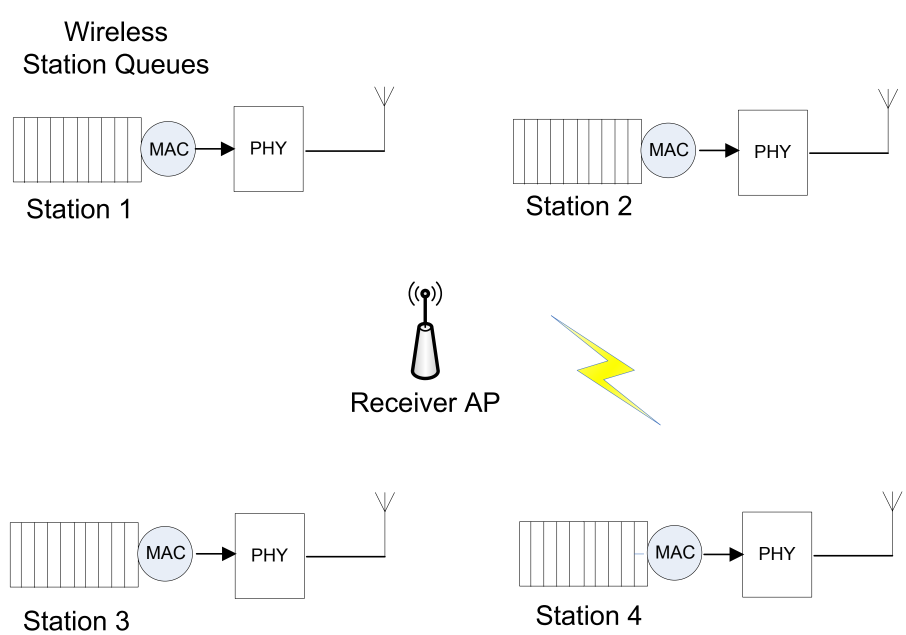

Fig. 5.8: Simulation Wireless Test-bed Configuration for Benchmarking Purpose

The loss rates and throughputs at layer 2 were evaluated for an increasing number of wireless stations where one queue per station has been implemented. The experimental testbed is shown in Fig. 5.8. The following configurations were used -

- 1 Access Point
- 2,3, and 4 stations
- Packet sizes - 512 B, 1024 B, 1500 B
- Each packet size has three different packet rates - 25 pps , 50 pps, 100 pps
- CBR Traffic

5.1.6.2 Benchmarking Results

Ethernet has an upper limit of 1500 B for packet sizes and hence this upper value. for packet size was chosen. Tests were also carried out for 300 pps and 500 pps packet rates but the stations reached saturation and hence they are not reported here. The throughput presented is the average throughput at the receiving stations for a particular packet rate and packet size for different number of stations. For example, if there are 4 different stations which each transmits 1500 B packets at a packet rate of 25 packets per second, the offered load is 1500 x 8 x 25 = 0.3 Mbps per station. Table 5.2(a) and Table 5.2(b): summaries the comparisons for NS-3 and the MAC simulator. There is a small discrepancy between results obtained through the simulator described in this work and results obtained from NS-3 simulations. It is believed that the failed frame retransmission (ReTx) feature is the reason for the apparent discrepancy. The ReTx feature is a default feature in the NS-3 but which has not been implemented in our simulator for these benchmarking tests. Per station based throughput and loss rate results can be found in Appendix

-C section. After comparing the throughput and loss rate results obtained from both the NS-3 and the MAC simulator, it can be concluded that the implemented MAC simulator provides good performance correlation with that of the NS-3.

Table 5.2(a): Average Throughput (in Mbps) Comparison for Different Number of Stations

		25 PPS			50 PPS			100 PPS		
STA	Packet Size	Avg. Offered Load	NS-3	MAC Simulator	Avg. Offered Load	NS-3	MAC Simulator	Avg. Offered Load	NS-3	MAC Simulator
4	1500 B	0.300	0.300	0.288	0.600	0.528	0.576	1.200	1.178	1.152
	1024 B	0.210	0.21	0.197	0.410	0.410	0.393	0.820	0.703	0.787
	512 B	0.100	0.100	0.098	0.210	0.210	0.197	0.410	0.376	0.373
3	1500 B	0.300	0.300	0.288	0.600	0.600	0.577	1.200	1.128	1.154
	1024 B	0.210	0.210	0.197	0.410	0.410	0.395	0.820	0.775	0.788
	512 B	0.100	0.100	0.098	0.210	0.210	0.187	0.410	0.410	0.374
2	1500 B	0.300	0.300	0.291	0.600	0.216	0.582	1.200	1.200	1.163
	1024 B	0.210	0.210	0.199	0.410	0.410	0.397	0.820	0.820	0.794
	512 B	0.100	0.100	0.094	0.210	0.210	0.189	0.410	0.388	0.378

Table 5.2(b): Average Loss Rate (%) Comparison for Different Number of Stations

		25 PPS		50 PPS		100 PPS	
STA	Packet Size	NS-3	MAC Simulator	NS-3	MAC Simulator	NS-3	MAC Simulator
4	1500 B	0.00	1.20	7.25	2.45	2.23	4.78
	1024 B	0.00	1.35	0.00	1.70	11.66	3.32
	512 B	0.00	0.18	0.00	1.35	3.40	3.68
3	1500 B	0.00	1.17	0.00	2.30	7.25	4.60
	1024 B	0.00	1.33	0.00	1.53	5.86	5.23
	512 B	0.00	0.17	0.00	2.30	0.00	3.60
2	1500 B	0.00	0.90	38.40	1.80	0.00	3.70
	1024 B	0.00	1.10	0.00	2.30	0.00	4.60
	512 B	0.00	0.60	0.00	2.10	2.19	3.20

5.2 Test Scenarios - Results and Analysis (Validation of the QoS Aware MPEG-4 Video Delivery Algorithm)

5.2.1 Downlink Configuration

The total simulation time was 300 seconds. As a 25 fps PAL video system has been implemented, it can be calculated that in 300 seconds there were 500 GOPs/station (or 500 x 15 = 7500 frames) present. Hence 7500 frames (or 500 GOPs) per station were simulated. It was interesting to notice that there was a difference between the theoretical GOP numbers (which is 500) and the GOP numbers obtained through *Perl* programming analysis of the real video clips (which ranges from 498 to 596 for different movies). It is believed that this discrepancy arises from deviations from the MPEG-4 standard within the FFMPEG application which was used to generate the video clips.

In every 25 fps PAL video clip there are 2 *I, 7 P* and *16 B* frames present every second. Hence the total BW requirement per second per station for an AVATAR clip is -

(2 x *I* Frame Size + 7 x *I* Frame Size + 16 x *B* Frame Size)

= (2 x 9952 + 7 x 6159 + 16 x 3832) bytes

= 124328.73 bytes or 994629 bits or 0.99 Mbps per video stream.

Hence for 5 stations containing 5 separate AVATAR clips the total load on the video queue is 4950 kbps or approximately 5 Mbps. Through simulation it has been observed that when the total load on the video queue and background queue were ~5 Mbps and ~3 Mbps (CBR) respectively the network would operate close to saturation. In the following section a detailed discussion regarding the AVATAR clip is presented as AVATAR has the largest average frame sizes of all 12 video clips. Results from all 12 different video clips are summarised in tables 5.6 – 5.9

5.2.1.1 AVATAR Clip Delivery

The 5 stations containing AVATAR clips in total had 7500 x 5 = 37500 frames which included 2500 *I* frames, 10000 *P* frames and 25000 *B* frames. They were sent to the video queue as depicted in the experimental setup of Fig. 5.3. CBR 1500 byte frame size background traffic (3 Mbps) was introduced in the background queue. It was noticed that in total 1936 video frames were lost due to collisions - the breakdown being 129 *I* frames (~ 5.2 %), 505 *P* frames (5.1 %) and 1302 (5.2 %) *B* frames (Fig. 5.9). Further analysis shows that –

Table 5.3: Breakdown of The Collided AVATAR Frames

<table>
<tr><th>Frame Type</th><th>Number of Collided Frames</th><th>Total</th><th>%</th></tr>
<tr><td>I</td><td>129</td><td>129</td><td>5.2</td></tr>
<tr><td>P4</td><td>127</td><td rowspan="4">505</td><td rowspan="4">5.1</td></tr>
<tr><td>P7</td><td>137</td></tr>
<tr><td>P10</td><td>133</td></tr>
<tr><td>P13</td><td>108</td></tr>
<tr><td>B2</td><td>129</td><td rowspan="10">1302</td><td rowspan="10">5.2</td></tr>
<tr><td>B3</td><td>118</td></tr>
<tr><td>B5</td><td>124</td></tr>
<tr><td>B6</td><td>135</td></tr>
<tr><td>B8</td><td>122</td></tr>
<tr><td>B9</td><td>134</td></tr>
<tr><td>B11</td><td>139</td></tr>
<tr><td>B12</td><td>146</td></tr>
<tr><td>B14</td><td>128</td></tr>
<tr><td>B15</td><td>127</td></tr>
</table>

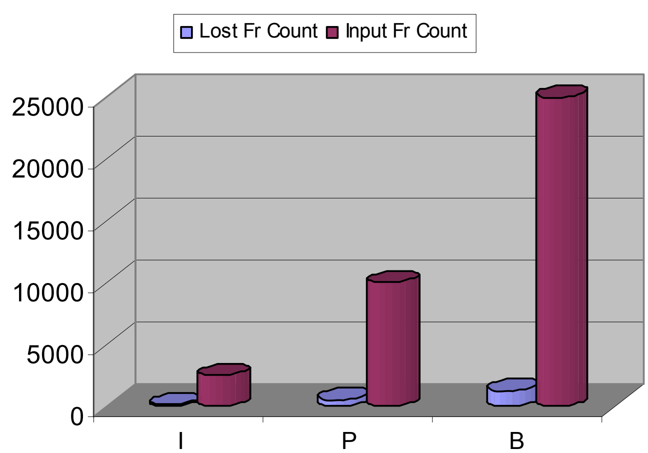

Fig. 5.9: Collided and Input AVATAR Frames (Lost Fr Count – Lost Frames, Input Fr Count – Number of Input Frames)

5.2.1.1.1 Bandwidth Loss Calculation for the *I* Frames

Table 5.4: AVATAR Frame Size

AVATAR	*I*	*P*	*B*
Frame Size	9952 bytes	6159 bytes	3832 bytes
	79616 bits	49272 bits	30656 bits
	~80 kbits	~49 kbits	~31 kbits

When an *I* frame was lost, there would be a severe impact on that particular GOP as bandwidth would be wasted in transmitting the remaining fourteen *P* and *B* frames. So for every lost *I* frame, the average wasted data for the fourteen *P* and *B* frames was –

(4 x 6159 + 10 x 3832) bytes

= 62956 bytes = 503648 bits = 504 kbits = 0.5 Mbits.

Hence the total data loss for an *I* frame was the data lost for the *I* frame and the wasted data for the *P* and *B* frames within the same GOP which were 80 kbits and 504 kbits respectively. For the 129 lost *I*- frames, the average *I*- frame only the bandwidth (BW) loss was –

(80 kbits x 129) / 300 sec = 35 kbps

The average wasted BW was (504 kbits x 129) / 300 sec = 218 kbps.

It was observed that in certain cases there was multiple frame loss within the same GOP. For example, *P* and *B* frames were lost within the same GOP which also lost its *I* frame. Hence taking this into account, i.e. avoiding double counting of these *P* and *B* frames, the total average wasted BW for lost *I* frames was calculated to be 204 kbps. Hence, the total BW lost for *I*-frame was (35 + 204) or 239 kbps.

5.2.1.1.2 Bandwidth Loss Calculation for the *P* Frames

The BW losses for four different *P* frames were:

(49 kbits x 127) / 300 sec = 21 kbps ………. for *P4* frames

(49 kbits x 137) / 300 sec = 22 kbps ………. for *P7* frames

(49 kbits x 133) / 300 sec = 22 kbps ………. for *P10* frames

(49 kbits x 108) / 300 sec = 18 kbps ………. for *P13* frames

For all 505 *P frames*, the BW lost is (49 x 505) / 300 = 83 kbps.

When one *P4, P7, P10 or P13* frame was lost, bandwidth was wasted in transmitting the remaining 11, 8, 5, and 2 *P* and *B* frames respectively within that GOP which would be of no use to the client. Hence the wasted BW for the *P* frames were -

For 127 *P4s:* (3 x 6159 + 8 x 3832) bytes = 49133 bytes = 393 kbits.

In kbps, (393 x 127) / 300 = 166 kbps

For 137 *P7*s*:* ((2 x 6159 + 6 x 3832) x 8 x 137) / (300 x 1000) kbps = 129 kbps

For 133 *P10*s*:* ((1 x 6159 + 4 x 3832) x 8 x 133) / (300 x 1000) kbps = 77 kbps

For 108 *P13*s*:* ((2 x 3832) x 8 x 108) / (300 x 1000) kbps = 22 kbps

It was observed that in certain cases *P* and *B* frames were lost within the same GOP which also lost its earlier *P* frame. Hence, the total average wasted BW for lost *P4, P7, P10 or P13* frames were calculated as 157, 124, 73 and 21 kbps respectively.

5.2.1.1.3 Bandwidth Loss Calculation for the *B* Frames

B frames depend on the previous and the next *I* or *P* frames to work. Hence the BW lost associated with the collided *B* frames was only for the loss in BW of *B* frames, i.e. there was no associated wasted BW.

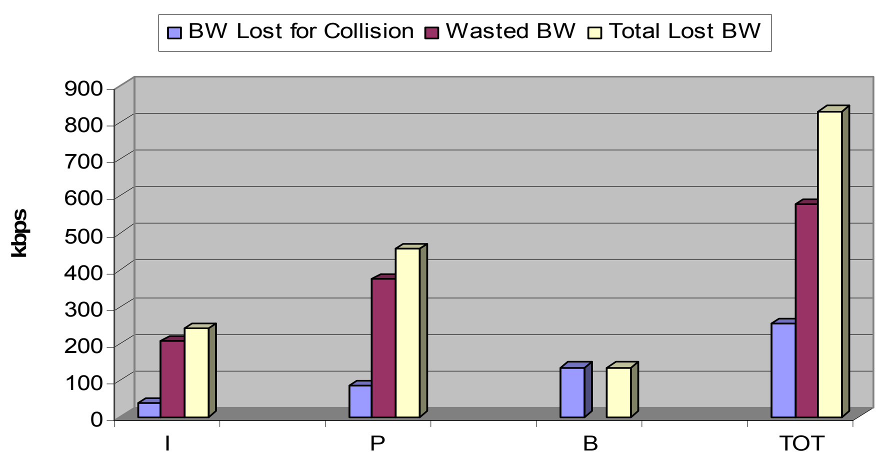

Fig. 5.10: BW Lost for Collision, Wasted BW, and Total Lost BW

A breakdown of the BW lost for different *B* frames was calculated as shown below-

For *B2,* (31 x 129)/ 300 = 13 kbps

For *B3,* (31 x 118)/ 300 = 12 kbps

For *B5,* (31 x 124)/ 300 = 13 kbps

For *B6,* (31 x 135)/ 300 = 14 kbps

For *B8,* (31 x 122)/ 300 = 13 kbps

For *B9,* (31 x 134)/ 300 = 14 kbps

For *B11,* (31 x 139)/ 300 = 14 kbps

For *B12,* (31 x 146)/ 300 = 15 kbps

For *B14,* (31 x 128)/ 300 = 13 kbps

For *B15,* (31 x 127)/ 300 = 13 kbps

For all 1302 *B* frames, (31 x 1302) / 300 = 134 kbps BW was lost. The BW losses due collided frames are shown in Fig 5.10 and summarised in the following table 5.5 –

Table 5.5: Summary of Bandwidth Loss

Frame Type	Number of Lost Frames	Frame only BW Lost (kbps)	Wasted BW (kbps)	Total BW Lost (kbps)	Lost	Wasted	Total BW Lost
I	**129**	**35**	**204**	**239**	**0.70%**	**4.10%**	**4.80%**
P4	127	21	157	178	0.42%	3.14%	3.56%
P7	137	22	124	146	0.44%	2.48%	2.92%
P10	133	22	73	95	0.44%	1.46%	1.90%
P13	108	18	21	39	0.36%	0.42%	0.78
B2	129	13	-	13	0.26%	0%	0.26%
B3	118	12	-	12	0.24%	0%	0.24%
B5	124	13	-	13	0.26%	0%	0.26%
B6	135	14	-	14	0.28%	0%	0.28%
B8	122	13	-	13	0.26%	0%	0.26%
B9	134	14	-	14	0.28%	0%	0.28%
B11	139	14	-	14	0.28%	0%	0.28%
B12	146	15	-	15	0.30%	0%	0.30%
B14	128	13	-	13	0.26%	0%	0.26%
B15	127	13	-	13	0.26%	0%	0.26%
All 4 *P* frames	**505**	**83**	**375**	**458**	**1.70%**	**7.60%**	**9.30%**
All 10 *B* frames	**1302**	**134**	**-**	**134**	**2.70%**	**0%**	**2.70%**
All *Frames*	**1936**	**252**	**579**	**831**	**5.10%**	**11.70%**	**16.80%**

It was seen that the loss rates of all three frame types (*I, P, B)* were approximately 5%. With this loss rate the probability of losing 1 frame out of 15 frames (one GOP) is –

$$n_{p_r} = {}^{15}p_1 = \binom{15}{1} p^1 (1-p)^{14} = \binom{15}{1} x 0.05^1 x (1-0.05)^{14} = 0.37 \quad \text{.............} (5.3)$$

i.e. there is approximately a one in three chance that the GOP will be corrupted, but this does not indicate the type of the frame.

Similarly, the probability of losing 2 and 3 frames out of 15 frames (1 GOP) are –

$$n_{p_r} = {}^{15}p_2 = \binom{15}{2} p^2 (1-p)^{13} = \binom{15}{2} x 0.05^2 x (1-0.05)^{13} = 0.13 \quad \text{.............} (5.4)$$

$$n_{p_r} = {}^{15}p_3 = \binom{15}{3} p^3 (1-p)^{12} = \binom{15}{3} x 0.05^3 x (1-0.05)^{12} = 0.03 \quad \text{.............} (5.5)$$

Hence there are respectively probabilities of 13% and 3% that 2 and 3 frames will be corrupted in a GOP for a loss rate of 5%.

The number of lost frames due to collisions could be minimized by effectively retransmitting them. From Table 5.5 it can be seen that a sizeable amount of BW was wasted for the earlier (*I, P4, P7,* etc.) collided frames as they render the rest of the frames within the same GOP undecodable for the client. So by applying the proposed QoS aware MPEG-4 video delivery algorithm if the lost frames were to be retransmitted, the total (including wasted) BW lost could be effectively minimized. For example, the *I* frames have a loss rate of around 5%. If they were allowed one retransmission opportunity then the bandwidth saved is greater than the bandwidth spent. The cost of saving 204 kbps would be 35 kbps with a 5% probability of loss which is a substantial amount of saving in terms of BW. In other words, the extra BW cost is being balanced by the significant amount of BW saving. The same could be argued for other frames as well.

It was seen that the loss rates of all three frame types (*I, P, B)*, in terms of frame count was approximately 5%. In terms of BW, 252 kbps (~ 5%) were lost due to collisions and 579 kbps (~12%) were wasted giving a total lost BW of 831 kbps (~17%) for a 4950 kbps input. If all the frames were to get at least one successful retransmission then it would cost an extra 252 kbps (~5%) in terms of BW on average across all the frames but it would produce significant BW saving (831 kbps or ~17%) and all the 500 GOPs could be delivered almost intact which would enhance the quality of the received video significantly. Hence there is a net BW saving of ~12%. Statistically, after one retransmission the probability of loss would be (0.05 x 0.05 = 0.0025) less than 1% which is the acceptable loss rate for streaming video according to the ITU-T standard [36,37].

Based on these findings, the simulator was modified to allow one retransmission for all three video frames. After analysing the obtained results, it was evident that one frame retransmission for all three frame types caused the system to have a loss rate of lower than 1% for *IPB* frames separately. The MAC exponential binary back-off mechanism results in a doubling of the CW, which helped to achieve the acceptable frame loss rate of $\leq 1\%$. Therefore it is evident that the reliability of the transmission attempts is improved and consequently the target loss rate was achieved at the same time minimizing the total BW lost thereby enhancing the video QoD. Results for all the 12 video clips are presented in tables 5.6 – 5.9 and are discussed in section 5.2.1.2.

Table 5.6: Number of Stations, Input Frame Count, Lost Frame Count (Without ReTx Being Applied)

	Num of STA Required for ~ 5 Mbps Load	*Input Frame Count*				*Without Retransmission, Lost Frame Count*							
		I	*P*	*B*	**Total**	*I*	*P*				*Total P*	*Total B*	**Total (All Frames)**
							P4	*P7*	*P10*	*P13*			
AVA	5	2500	10000	25000	37500	129	127	137	133	108	505	1302	1936
						5.2%					5.1%	5.2%	5.2%
2012	8	4000	16000	40000	60000	212	224	222	202	204	852	2234	3298
						5.3%					5.3%	5.6%	5.5%
DH	8	4000	16000	40000	60000	222	194	212	252	225	883	2164	3269
						5.6%					5.5%	5.4%	5.4%
KA	6	3000	12000	30000	45000	136	146	166	153	158	623	1559	2318
						4.5%					5.2%	5.2%	5.2%
LK	12	6000	24000	60000	90000	307	259	338	330	343	1270	3210	4787
						5.1%					5.3%	5.4%	5.3%
IA	12	6000	24000	60000	90000	309	233	316	327	353	1229	3286	4824
						5.2%					5.1%	5.5%	5.4%
RUG	10	5000	20000	50000	75000	293	299	298	293	283	1173	2900	4366
						5.9%					5.9%	5.8%	5.8%
FB	12	6000	24000	60000	90000	321	210	342	312	352	1216	3300	4837
						5.4%					5.1%	5.5%	5.4%
BBC	11	5500	22000	55000	82500	309	321	321	304	308	1254	3189	4752
						5.6%					5.7%	5.8%	5.8%
ANT	15	7500	30000	75000	112500	405	195	403	379	368	1345	3515	5265
						5.4%					4.5%	4.7%	4.7%
MD	17	8500	34000	85000	127500	257	384	412	436	406	1638	3852	5747
						3.0%					4.8%	4.5%	4.5%
MZ	17	8500	34000	85000	127500	235	365	403	435	373	1576	3975	5786
						2.8%					4.6%	4.7%	4.5%

Table 5.7: Number of Stations, Input Frame Count, Lost Frame Count (With 1 ReTx For All Types of Failed Frames Due to MAC Collisions)

	Num of STA Required for ~ 5 Mbps Load	Input Frame Count				With 1 Retransmission, Lost Frame Count							
		I	*P*	*B*	Total	*I*	*P*				*Total P*	*Total B*	Total (All Frames)
							P4	*P7*	*P10*	*P13*			
AVA	5	2500	10000	25000	37500	1 < 1%	2	3	0	1	6 < 1%	18 < 1%	25 < 1%
2012	8	4000	16000	40000	60000	0 0%	1	2	5	0	8 < 1%	19 < 1%	27 < 1%
DH	8	4000	16000	40000	60000	0 < 1%	0	2	4	3	9 < 1%	16 < 1%	25 < 1%
KA	6	3000	12000	30000	45000	1 < 1%	3	0	4	4	11 < 1%	14 < 1%	26 < 1%
LK	12	6000	24000	60000	90000	2 < 1%	2	2	2	1	7 < 1%	9 < 1%	18 < 1%
IA	12	6000	24000	60000	90000	1 < 1%	2	2	2	1	7 < 1%	15 < 1%	23 < 1%
RUG	10	5000	20000	50000	75000	3 < 1%	2	3	0	1	6 < 1%	12 < 1%	21 < 1%
FB	12	6000	24000	60000	90000	1 < 1%	1	2	3	2	8 < 1%	17 < 1%	26 < 1%
BBC	11	5500	22000	55000	82500	2 < 1%	3	4	0	0	7 < 1%	14 < 1%	23 < 1%
ANT	15	7500	30000	75000	112500	1 < 1%	0	2	1	1	4 < 1%	10 < 1%	15 < 1%
MD	17	8500	34000	85000	127500	0 < 1%	0	2	1	0	3 < 1%	16 < 1%	19 < 1%
MZ	17	8500	34000	85000	127500	0 < 1%	0	1	0	1	2 < 1%	20 < 1%	22 < 1%

Table 5.8: Total Input Load, BW Lost Due to Collision and Waste, Lost Frame Count (Without Any ReTx)
C = BW Lost Due to Collision Only, W = Wasted BW

	Num of STA Required for ~ 5 Mbps Load	Input Load Per STA (Mbps)	Total Input Load (Mbps)	With 1 Retransmission, BW Lost(kbps)																		
				I			*P4*			*P7*			*P10*			*P13*			*All P*	*All B*	Total Lost (All Frames)	Total Lost (All Frames) (%)
				C	W	C+W	C	W	C+W	C	W	C+W	C	W	C+W	C	W	C+W				
AVA	5	0.99	4.97	35	204	239	21	157	178	22	124	146	22	73	95	18	21	39	458	134	831	16.80%
2012	8	0.66	5.28	41	222	264	25	182	207	25	129	153	22	71	94	23	25	48	501	144	909	17.22%
DH	8	0.62	4.96	43	217	260	19	147	166	21	115	136	25	83	108	22	26	49	459	134	853	17.17%
KA	6	0.83	4.98	27	188	215	20	157	177	23	128	151	21	71	92	22	26	48	468	138	821	16.42%
LK	12	0.43	5.16	54	193	247	19	126	145	25	117	141	24	68	92	25	24	49	427	117	791	15.22%
IA	12	0.43	5.16	57	188	244	15	110	126	21	106	127	22	66	88	23	24	48	388	120	753	14.76%
RUG	10	0.52	5.20	52	234	285	29	186	214	29	131	159	28	77	105	27	24	51	530	130	945	18.08%
FB	12	0.42	5.04	68	182	250	14	92	106	23	106	129	21	57	79	24	21	45	359	105	714	14.07%
BBC	11	0.44	4.84	63	191	254	22	154	176	22	109	131	21	62	83	21	21	43	433	117	804	16.52%
ANT	15	0.33	4.95	85	159	245	9	59	68	18	87	105	17	49	67	17	16	32	272	80	597	12.04%
MD	17	0.29	4.93	52	83	136	13	97	110	14	74	88	15	47	62	14	15	29	290	75	501	10.17%
MZ	17	0.29	4.93	50	74	123	14	89	103	15	70	85	16	45	61	14	13	27	275	71	469	9.54%

Table 5.9: Total Input Load, BW Lost Due to Collision and Waste, Lost Frame Count (With 1 ReTx for all Types of Failed Frames Due to MAC Collisions) C = BW Lost Due to Collision Only, W = Wasted BW

	Num of STA Required for ~ 5 Mbps Load	Input Load Per STA (Mbps)	Total Input Load (Mbps)	With 1 Retransmission, BW Lost(kbps)																		
				I			*P4*			*P7*			*P10*			*P13*			*All P*	*All B*	Total Lost (All Frames)	Total Lost (All Frames) (%)
				C	W	C+W	C	W	C+W	C	W	C+W	C	W	C+W	C	W	C+W				
AVA	5	0.99	4.97	0.3	1.73	2.03	0.35	2.7	3.05	0.5	3	3.5	0	0	0	0.17	0.22	0.39	7	1.9	11	< 1%
2012	8	0.66	5.28	0	0	0	0.11	0.84	0.95	0.22	1.24	1.46	0.55	1.85	2.4	0	0	0	4.81	1.21	6.02	< 1%
DH	8	0.62	4.96	0	0	0	0	0	0	0.2	1.16	1.36	0.4	1.42	1.82	0.3	0.4	0.7	3.88	1.02	4.9	< 1%
KA	6	0.83	4.98	0.2	1.45	1.65	0.41	3.4	3.81	0	0	0	0.55	2	2.55	0.55	0.93	1.48	7.84	1.26	10.75	< 1%
LK	12	0.43	5.16	0.35	1.4	1.75	0.15	1.04	1.19	0.15	0.74	0.89	0.15	0.3	0.45	0.08	0.08	0.16	2.69	0.33	4.77	< 1%
IA	12	0.43	5.16	0.2	0.7	0.9	0.15	1.04	1.19	0.15	0.74	0.89	0.15	0.45	0.6	0.08	0.08	0.16	2.84	0.55	4.29	< 1%
RUG	10	0.52	5.20	0.53	2.49	3.02	0.19	1.3	1.49	0.29	1.38	1.67	0	0	0	0.1	0.09	0.19	3.35	0.52	6.89	< 1%
FB	12	0.42	5.04	0.21	0.6	0.81	0.07	0.47	0.54	0.13	0.67	0.8	0.20	0.60	0.8	0.13	0.13	0.26	2.4	0.57	3.78	< 1%
BBC	11	0.44	4.84	0.41	1.3	1.71	0.21	1.51	1.72	0.28	1.44	1.72	0	0	0	0	0	0	3.44	0.51	5.66	< 1%
ANT	15	0.33	4.95	0.21	0.42	0.63	0	0	0	0.09	0.47	0.56	0.05	0.14	0.19	0.05	0.05	0.1	0.85	0.23	1.71	< 1%
MD	17	0.29	4.93	0	0	0	0	0	0	0.07	0.38	0.45	0.03	0.11	0.14	0	0	0	0.59	0.32	0.91	< 1%
MZ	17	0.29	4.93	0	0	0	0	0	0	0.04	0.18	0.22	0	0	0	0.04	0.03	0.07	0.29	0.33	0.62	< 1%

5.2.1.2 Discussion Regarding All the 12 Video Clips on Frames Loss Rate and Total Lost Bandwidth

1. Due to differences in frame sizes, different numbers of video streams were required for generating a target throughput (i.e. offered load) of approximately 5 Mbps for different video contents for the video queue. For example, AVATAR and MZ have the largest and smallest average frame sizes respectively. As a result 5 AVATAR and 17 MZ video streams (1 stream per station) were required to generate ~5 Mbps offered load on the video queue of the AP (Table 5.6) As different video clips have different frame sizes the capacity perceived by the video queue and the system would be completely different although the offered load is the same for different videos.

2. For a 300 second simulation run, each video stream has 500 GOPs which translates to 500 x 15 or 7500 video frames (i.e. 500 *I* frame, 2000 *P* frames and 10000 *B* frames). Hence 5 AVATAR video streams have 7500 x 5 = 37500 frames which includes 2500 *I* frames, 10000 *P* frames and 25000 *B* frames. Similarly 17 MZ stations sends 7500 x 17 = 127500 frames (8500 *I* frames, 34000 *P* frames and 85000 *B* frames) to the video queue over a 300 second simulation period (Table 5.7).

3. The simulated AP has two contending queues – one video and one background queue. After the MAC simulation was run for 300 seconds, it

was calculated (Table 5.8) that without employing failed frame ReTx due to MAC collisions, the loss rates were approximately 5% for *IPB* frames separately. 1 ReTx for all frames was sufficient to reduce the frames loss rates from ~ 5% to the target $\leq$1% for all 12 video clips (Table 5.9) which suggests that more frames could be delivered successfully.

4. Without the failed frame ReTx, the total bandwidth lost (loss due to collisions and wastage due to frame hierarchy) is ~10-18% for the 12 video clips (Table 5.8). However, when 1 frame ReTx was applied the percentage of lost bandwidth reduced significantly and the net savings in bandwidth were observed in the range of ~9 -17% for all 12 clips (Table 5.9).

5.2.2 Uplink Configuration

In order to validate the proposed QoS aware MPEG-4 video delivery algorithm an uplink network configuration was simulated in *C* programming language as depicted in Fig. 5.4. In this network topology several wireless stations were contending to access the medium. Each MPEG-4 video or CBR background data (frame size 1500 bytes) sending station had one queue each and the station that won the transmission opportunity sends data to the receiver AP. The simulation was run for 300 seconds. The proposed QoS aware MPEG-4 video delivery algorithm would be implemented in each of the MPEG-4 video server's queues. Compared to the downlink case which was described in section 5.5.1, a higher contention was present in the uplink case as the number of stations contending to access the medium was greater.

Tests were completed using the developed uplink simulator and the obtained data were analysed in a systemic manner for all 12 video clips. Only results for AVATAR and MZ video clips will be discussed in detail as they have the largest and smallest average frame sizes respectively of all the 12 video clips. All the test results of the 12 clips are summarised in Table 5.12.

5.2.2.1 Analysis of AVATAR (AVA) and Mark Zuckerberg (MZ) Video Clips

5.2.2.1.1 Video Capacity

In general the capacity of the WLAN is directly proportional to the size of the frames and inversely related to station contention. The buffer occupancy can be defined as the occupancy of the MAC transmit buffer just after each MAC transmission attempt. A station is in saturation when its buffer always has at least one frame to transmit, i.e. at saturation a station's buffer is never empty and hence the station is always contending for accessing the wireless medium. When the average buffer occupancy starts increasing the probability of buffer overflow increases. Buffer overflow occurs when there is insufficient capacity in the transmit buffer to accommodate the arrival of new packets to be transmitted. Whenever the average buffer occupancy is greater than zero then there is a finite probability of buffer overflow. The "Video Capacity" can be defined as the maximum number of video streams capable of being accommodated in the system which gives rise to zero buffer occupancies for all contending video stations queues. The MAC model written in *C* programming language was used to determine the maximum number of video streams that does not lead the system into saturation.

Simulation results show that for both AVA and MZ clips the Video Capacity of the system was 2 video streams. For this configuration the video stations buffer occupancies were always zero. If the number of contending video stations increases then video frames start accumulating in the video buffers. So system capacity in terms of throughput may differ from one content type to another but is the same for both content types (i.e. it is largely independent of the content of the streams). The following four figures (Fig. 5.11(a) – Fig. 5.11(d)) show the buffer occupancies over time of the contending MZ video clips for different numbers of contending video stations -

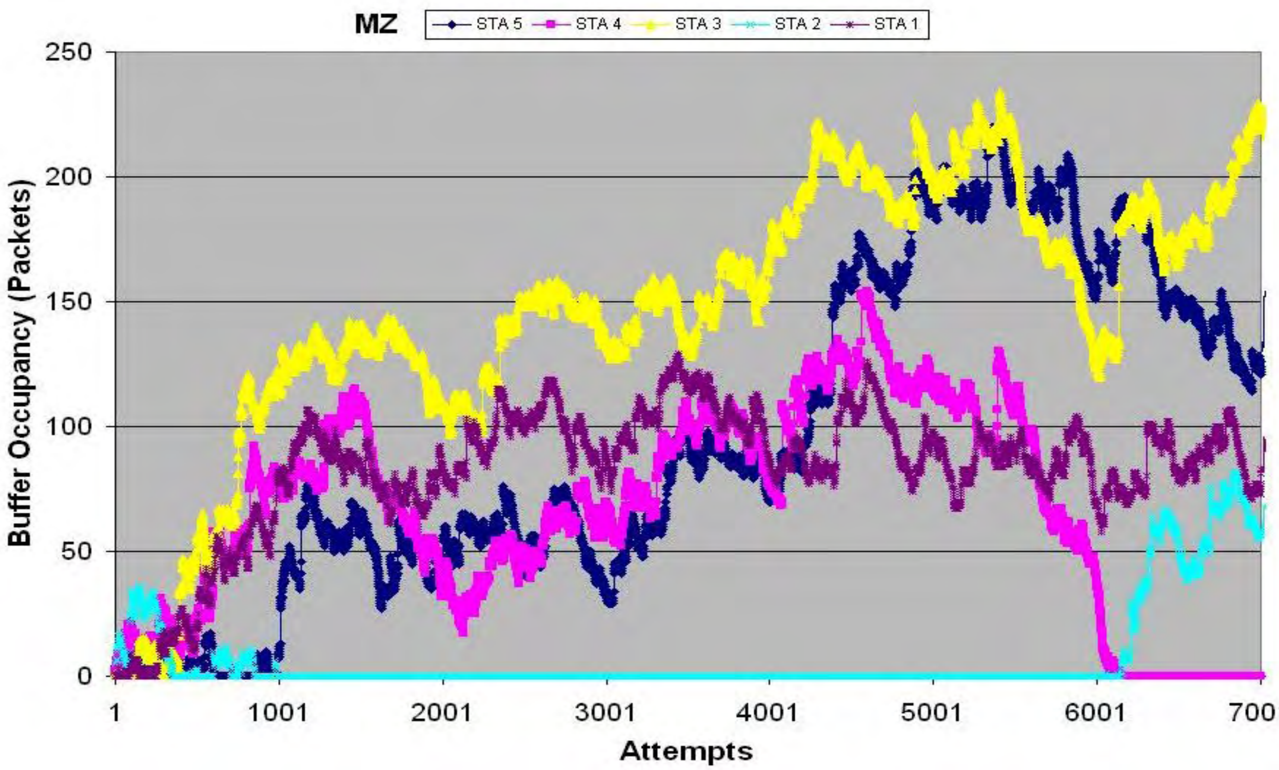

Fig. 5.11(a): Buffer Occupancies of 5 Stations for MZ Clip

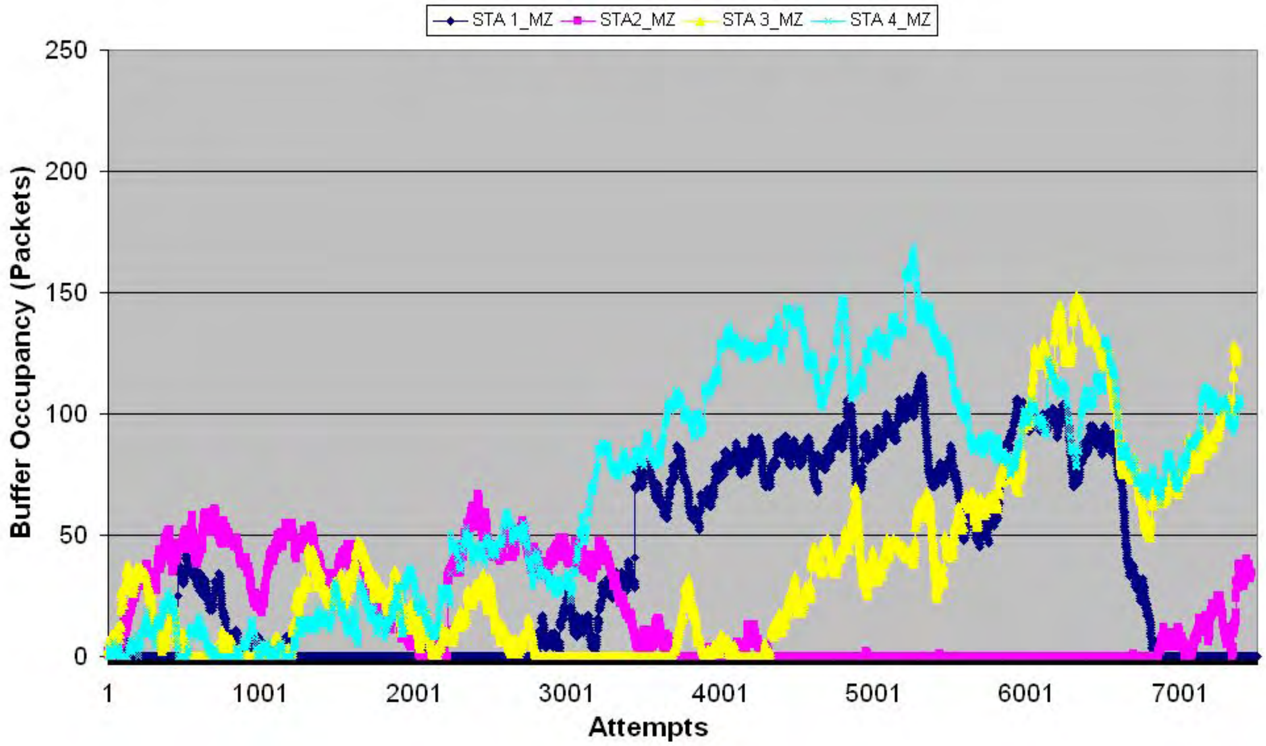

Fig. 5.11(b): Buffer Occupancies of 4 Stations for MZ Clip

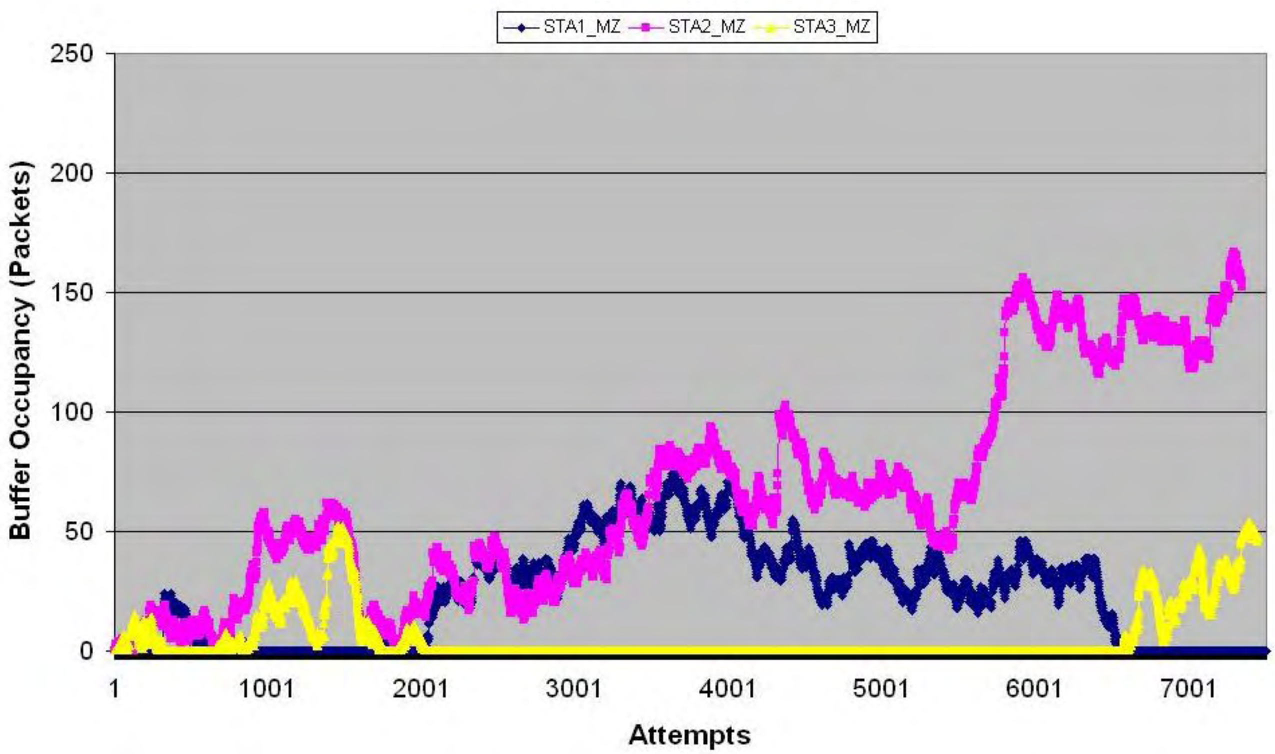

Fig. 5.11(c): Buffer Occupancies of 3 Stations for MZ Clip

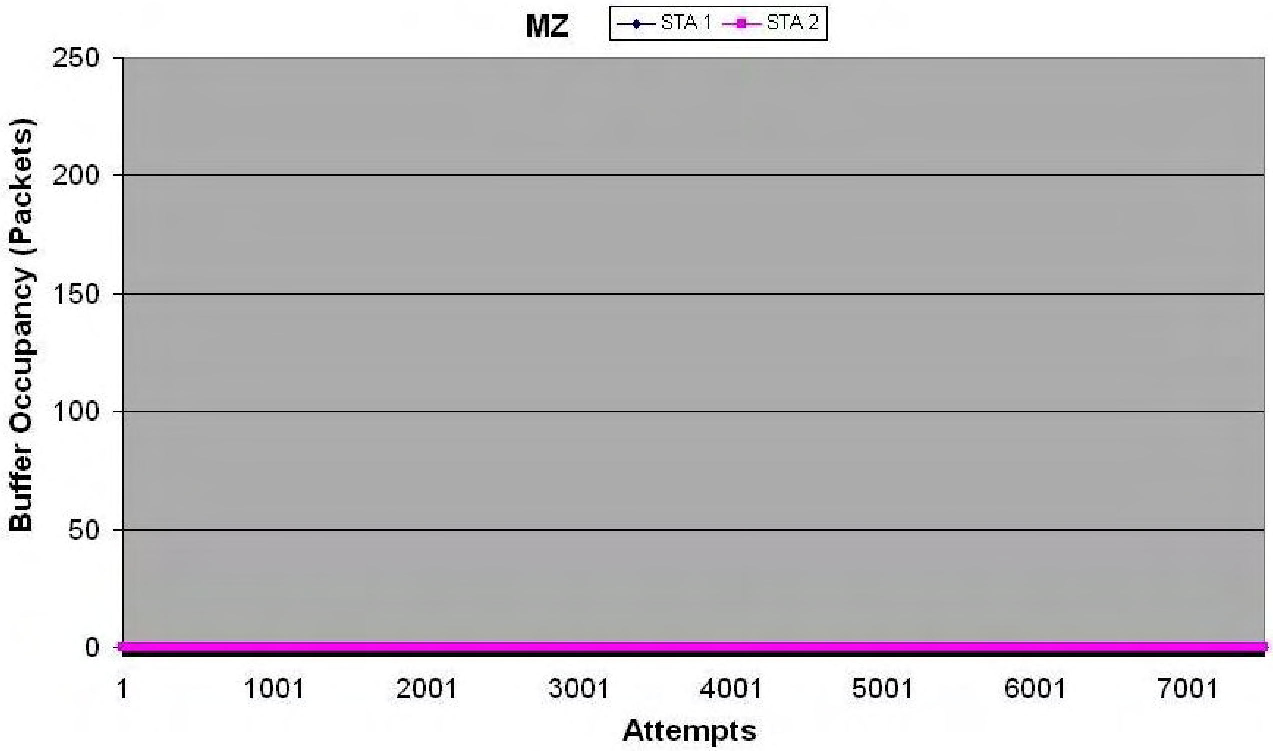

Fig. 5.11(d): Buffer Occupancies of 2 Stations for MZ Clip

5.2.2.1.2 Effect of CBR Background Traffic

CBR background traffic was generated by a third station (i.e. 2 video and 1 contending background station) to measure the effect on the network. The goal was to determine the level of background load where the system would fail, i.e. when the video buffer occupancies would be greater than zero (i.e. there is always one packet to transmit). It was found that 2 AVA or MZ video streams could tolerate a maximum of ~550 kbps of background (using 1500 byte size packets) load, i.e. if a station carrying more than ~550 kbps contends with the 2 AVA stations, the average buffer occupancy in the two video stations would be non zero. If there was one video station carrying AVA or MZ clip and 1 CBR background station present in the network contending for access, it was found that for a maximum of ~2.4 Mbps background load the video station's buffer occupancy remain zero at all time. The levels of tolerable background loads for 2 and 1 streams of AVA and MZ clips are summarised in table 5.12 -

Table 5.10: Maximum Tolerable Background Load for Different Number of Video Streams

Video Clips	2 Video Streams	1 Video Stream
AVA	~550 kbps	~2.4 Mbps
MZ	~500 kbps	~2.2 Mbps

5.2.2.1.3 Implementation of the Proposed QoS Aware MPEG-4 Video Delivery Algorithm

The strategy for implementing the proposed QoS aware MPEG-4 video delivery algorithm described earlier can be shown in the following flowchart (Fig. 5.12) –

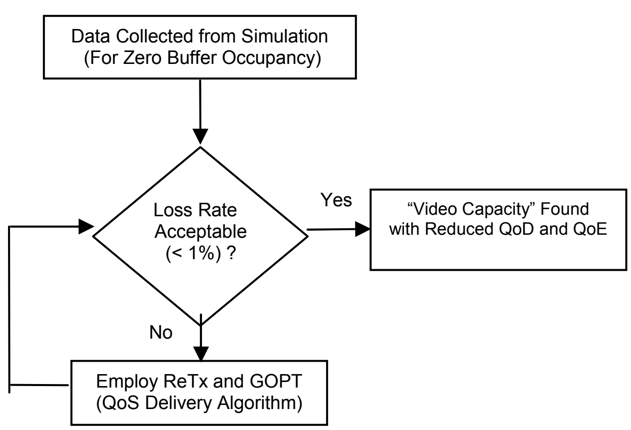

Fig. 5.12: QoS Aware MPEG-4 Video Delivery Algorithm Implementation for the Uplink Scenario

Data Collected from Simulation (For Zero Buffer Occupancy)

After establishing the system's video capacity in terms of the number of video streams and maximum tolerable background load for the MPEG-4 streams over WLANs, video frame loss rates (in terms of percentages for the *I, P,* and *B* frames) were calculated by effectively tracking every video frame from the traffic generator until its transmission (both successful and failed). Thus the QoD of the video frames was analysed. Using *Perl* scripts it was calculated that the frames loss rates for both AVA and MZ clips were more than the acceptable level (of 1%) for all three frame types.

Loss Rate Acceptable (< 1%)?

At first, the ReTx mechanism was used in the simulator to achieve the target $\leq$1% loss rate for all three frame types. As explained in section 5.3.1, failed frame retransmission (ReTx) facilitates the management of transmission losses due to MAC contention thereby increasing the QoD at the expense of a higher probability of buffer overflow by reducing the effective average buffer service rate.

Employ ReTx and GOPT (QoS Aware MPEG-4 Video Delivery Algorithm)

Simulation results indicate that allowing for 1 ReTx attempt for all three frames types was sufficient to achieve the target $\leq$1% loss rate but it results in a non zero probability of buffer overflow. To mitigate the additional bandwidth requirement, the

GOPT mechanism was employed. Determining the level of the GOPT required achieving the earlier calculated video capacity under background load conditions to eliminate the probability of buffer overflow, i.e. achieving zero buffer occupancies in the video queues. The average buffer frame arrival rate is decreased by GOPT thereby eliminating the probability of buffer overflow at the expense of a reduced QoE. Hence “Video Capacity” with reduced QoD and QoE is obtained.

It was calculated that for the AVA clip, 2 video and 1 background stations scenario, 40% GOPT (i.e. last 6 frames - $P_{10}B_{11}B_{12}\ P_{13}B_{14}B_{15}$ - discarded out of 15 frames in every GOP) was required in addition to 1 frame ReTx to obtain zero buffer occupancies in both the video queues. When 1 video station carrying the AVA clip was contending with 1 background station, 1 frame ReTx and 20% GOPT (i.e. last 3 frames - $P_{13}B_{14}B_{15}$ - discarded out of 15 frames in every GOP) were required to obtain the same target. For the MZ clip the frame ReTx level was the same but a greater level of GOPT was required. The corresponding results are summarised in the following tables 5.11(a) – 5.11(d).

Table 5.11(a): AVA: Frames Loss Rate in the 2 Stations Containing Video Streams			
2 Video Streams + 1 BAK Stream	*I* Frame	*P* Frame	*B* Frame
No ReTx 15 Frames (No GOPT)	~ 4.7%	~7%	~ 6.9%
1 ReTx 15 Frames (No GOPT)	~ 0.9%	~ 0.9%	~ 0.7%
1 ReTx 9 Frames (40% GOPT)	~ 0.6%	~ 0.2%	~ 0.6%

Table 5.11(b): AVA: Frames Loss Rates in the 1 Station Containing Video Streams			
1 Video Streams + 1 BAK Stream	*I* Frame	*P* Frame	*B* Frame
No ReTx 15 Frames (No GOPT)	~ 2.7%	~4.4%	~ 4.58%
1 ReTx First 12 Frames (20% GOPT)	~ 0.4%	~ 0.2%	~ 0.2%

Table 5.11(c): MZ: Frames Loss Rate in the 2 Stations Containing Video Streams			
2 Video Streams + 1 BAK Stream	*I* Frame	*P* Frame	*B* Frame
No ReTx 15 Frames (No GOPT)	~ 4.7%	~7%	~ 6.9%
1 ReTx 15 Frames (No GOPT)	~ 0.9%	~ 0.9%	~ 0.74%
1 ReTx First 6 Frames (60% GOPT)	~ 0.5%	~ 0.2%	~ 0.3%

Table 5.11(d): MZ: Frames Loss Rates in the 1 Station Containing Video Streams			
1 Video Streams + 1 BAK Stream	*I* Frame	*P* Frame	*B* Frame
No ReTx 15 Frames (No GOPT)	~ 3.6%	~2.2%	~ 2.2%
1 ReTx First 9 Frames (40% GOPT)	~ 0.4%	~ 0.1%	~ 0.12%

As mentioned earlier, the capacity of a WLAN system is proportional to frame sizes and inversely related to the contention present in the medium. A network that has 2 video and 1 background stations, experiences a greater contention than a network having 1 video and 1 background stations. Hence, a higher level of GOPT is required when contention is greater as evident by the results shown in the tables 5.11(a) – 5.11(d) above. Analysis of the simulation results show that the required level of GOPT was quite high (sometimes 60%). This was due to the fact that a very conservative approach was adopted in designing and implementing the QoS aware MPEG-4 video delivery algorithm, i.e. setting an average buffer occupancy target of zero for video stations queues.

Video Quality

In terms of QoS, the MAC simulator realised in this work provides QoD information (in terms of loss rate) only. However, the final arbiter of quality is the end user which is represented by QoE. A widely used QoE metric is the *PSNR.* The quality of the videos was evaluated for the different GOPT levels and loss rates obtained through simulation for different clips. The following *PSNR* estimation formula, described in [155] has been used for this purpose–

$$PSNR = 20\log_{10}\left(\frac{MAX\ Bitrate}{\sqrt{(EXP\ Thr - CRT\ Thr)^2}}\right) \quad (5.6)$$

Where,

PSNR = Peak Signal to Noise Ratio

MAX Bitrate = Average bit rate of the multimedia stream resulting from the encoding process

EXP Thr = Expected average throughput

CRT Thr = Actual Throughput

Detail calculation for AVATAR clip is shown below –

It has been calculated that the average sizes of *I, P, B* frames for AVATAR are 9952, 6159, and 3832 bytes respectively. As there are 2 *I*, 7 *P*, and 16 *B* (i.e. 25

fps) frames present per second, it gives a bit rate of 0.99 Mbps (i.e. *MAX Bitrate* = 1 Mbps). When 20% GOPT is applied, there are 2 *I*, 5 *P*, and 13 *B* frames per second present in the video stream. Hence the expected throughput is 0.8 Mbps. If 40% GOPT is applied, there are 2 *I*, 3*P*, and 10 *B* frames per second present with an expected throughput of 0.61 Mbps. The average loss rates for all three frames have been shown in table 5.11 are 0.2% and 0.5% for 20% and 40% GOPT levels respectively. Taking into account of these loss rates, the *PSNR* values for 20% and 40% GOPT levels could be calculated -

$$PSNR = 20\log_{10}\left(\frac{0.99}{\sqrt{(0.8 - 0.7984)^2}}\right) = 55.83dB \text{, at 20\% GOPT}$$

$$PSNR = 20\log_{10}\left(\frac{0.99}{\sqrt{(0.61 - 0.6069)^2}}\right) = 50.22dB \text{, at 40\% GOPT}$$

The following table (Table 5.12) summarises the level of GOPT required and corresponding video quality results for different content types-

Table 5.12: Level of GOPT required in Addition to 1 Frame ReTx for all 12 Different Clips for Target Zero Buffer Occupancies

Genre	Content Type	BitRate (Mbps)	Average Frame Size (byte)	Level of GOPT* Required for Different Network Topologies and Corresponding PSNR values in dB			
				2 Video and 1 BAK Stations		1 Video and 1 BAK Stations	
				GOPT Level	PSNR (dB)	GOPT Level	PSNR (dB)
CGI	AVA	0.99	6648	40%	50.22	20%	55.83
	2012	0.66	4633	40%	47.22	20%	52.28
Action	DH	0.62	4420	40%	48.47	20%	52.24
	KA	0.83	5304	40%	48.72	20%	55.86
Action	LK	0.43	3561	60%	52.11	40%	54.24
	IA	0.43	3575	60%	51.92	40%	54.10
Sport	RUG	0.52	3965	60%	51.05	40%	58.06
	FB	0.42	3951	60%	51.47	40%	57.43
Documentary	BBC	0.44	3851	60%	50.15	40%	54.01
	ANT	0.33	3486	60%	52.57	40%	62.95
Talking Head	MD	0.29	3233	60%	54.66	40%	62.69
	MZ	0.29	3321	60%	54.55	40%	62.66

[* As explained earlier, 20%, 40%, and 60% GOPT mean that the last 3 frames ($P_{13}B_{14}B_{15}$), 6 frames ($P_{10}B_{11}B_{12}$ $P_{13}B_{14}B_{15}$) and 9 frames ($P_7B_8B_9$ $P_{10}B_{11}B_{12}$ $P_{13}B_{14}B_{15}$) are discarded respectively from every GOP (IB_2B_3 $P_4B_5B_6$ $P_7B_8B_9$ $P_{10}B_{11}B_{12}$ $P_{13}B_{14}B_{15}$).**]**

As AVA clip frames are larger in size compared to all other clips' frame sizes, when it gets a transmission opportunity it holds the medium for a longer time. But as MZ frames are smaller compared to AVA frames, background traffic wins more share of the bandwidth. Larger frames compete with background traffic more efficiently in seizing network bandwidth than smaller frames. This is the reason that the level of GOPT for AVA is smaller than that of MZ for the same number of video and background stations due to the contention based nature of the WLANs. Similar trend is observed for other clips as well, i.e. experimental results show that video clips with larger average frame sizes require comparatively less level of GOPT for both network topologies.

The video quality of the clips was presented in Table 5.12 in terms of *PSNR* values which is a full reference objective metric. This metric has been discussed in detail in chapter 2. The higher the *PSNR* value the better the quality. The maximum theoretical achievable *PSNR* value is 100 dB. For lossy image and video compression cases typical values of PSNR are between 30 and 50 dB while, minimum *PSNR* value for acceptable wireless video transmission is 20 dB [156, 157]. The video quality results of Table 5.12 indicate that the more the level of GOPT the less the *PSNR*, i.e. when the GOPT level increases (i.e. more video frames are discarded) the video quality decreases. The use of the *PSNR* metric provides a measure of the impact of GOPT on QoE. It would be expected that the high action movies (e.g. AVATAR, 2012, Die Hard etc.) would be impacted more by the GOPT than the low action talking head video clips because they are inherently different . in their spatio-temporal characteristics. But Table 5.12 shows similar levels of reduction in *PSNR* values (around 5 to 8 dB) for all the clips the reason being all the videos have same numbers of *I, P, B* frames implemented over 300 seconds i.e. as simulated video frames were used in the cases described here. This would suggest that the *PSNR* metric is not suited to measure the impact of GOPT on the end user experience. Hence it is suggested that live trials might be employed to assess the video quality after implementing GOPT.

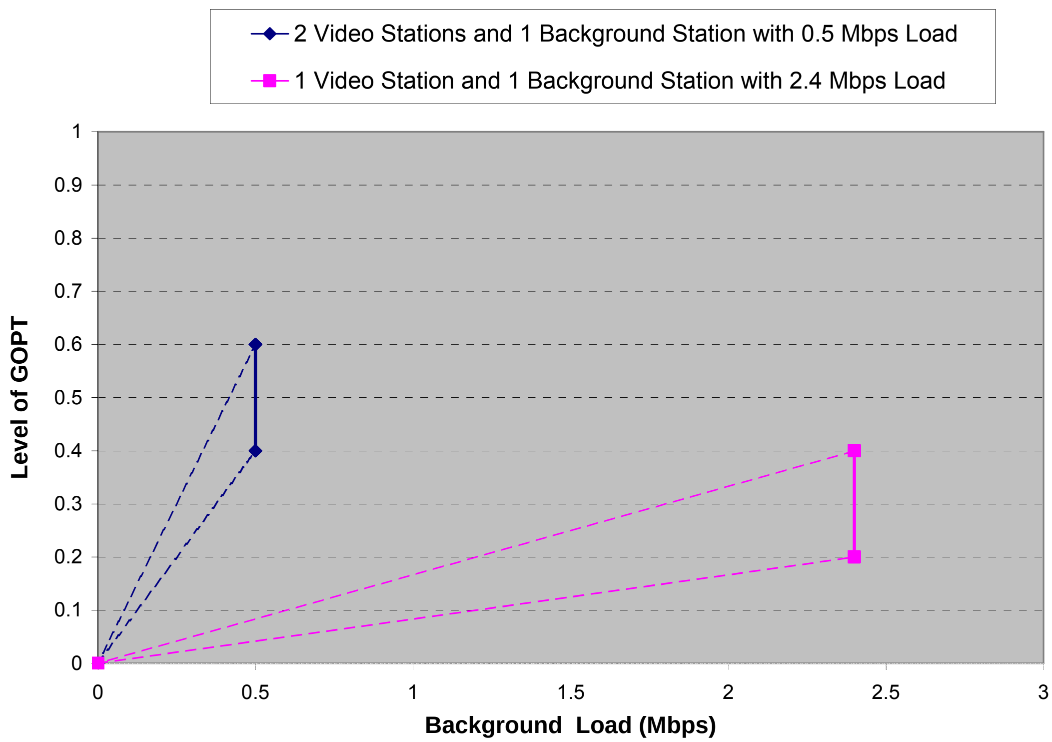

Fig. 5.13: Demonstration of Different Levels of GOPT Requirement for Different Network Topologies to Obtain Zero Buffer Occupancies in Addition to 1 ReTx for All Frames

The experimental results of Table 5.12 are also presented in Fig. 5.13 where it is evident that for the 2 video stations and 1 background station (0.5 Mbps) scenario, the maximum and minimum level of GOPT required are 60% and 40% respectively for all 12 test clips in addition to 1 ReTx for all frames to obtain zero buffer occupancies. Less amount of GOPT is required for the 1 video station and 1 background station (2.4 Mbps) topology.

Based on the simulation results and analysis, it can be concluded that the proposed QoS aware MPEG-4 video delivery algorithm involving frame ReTx and GOPT is effective in improving QoD of streamed video over IEEE 802.11b WLANs.

5.3 Benefits of the Proposed QoS Aware MPEG-4 Video Delivery Algorithm

a) The proposed novel QoS aware MPEG-4 video delivery algorithm describes the streaming of MPEG-4 video over a WLAN by proposing an alternative use of the QoS mechanisms provided for under the IEEE 802.11 standards that can guarantee a QoS performance improvement for video applications.

b) It is simple in its implementation and yet highly effective. Although it is proposed for IEEE 802.11b WLANs, it is generic in nature. As it is concerned with buffer occupancy; it would work with all types of IEEE 802.11 based WLANs (e.g. b/g/a/n). It is applicable for a wide range of video content (e.g. H.263/.264) other than the MPEG-4 format which can be segregated into their constituent *IPB* frames.

c) It employs two mechanisms only, namely failed frame ReTx and GOPT and is based on the measurement of the buffer occupancy metric which is measurable through implementing the MAC buffers for the IEEE 802.11 WLANs. The ReTx mechanism effectively increases the QoS by minimizing the transmission losses at the expense of an increased buffer overflow probability. It reduces bandwidth wastage by effectively retransmitting the failed frames. The GOPT mechanism reduces the probability of buffer overflow at the expense of a reduced QoE. Thus the QoS aware MPEG-4 video delivery algorithm aims to achieve an optimal trade off between these two mechanisms in order to eliminate buffer overflow and minimise transmission losses.

d) The algorithm aims to replace uncontrolled packet loss due to buffer overflow and MAC collisions by a controlled prioritized packet loss scheme that permits a graceful degradation in QoD for MPEG-4 video streamed over IEEE 802.11b networks. It achieves the ITU-T target specified for loss rate of streamed video transmission. This ensures the realisation of the most favourable network conditions for the delivery of MPEG-4 video frames on WLANs.

e) Through extensive simulations it has been shown to provide a significant improvement in the QoS performance for video streaming applications for both uplink and downlink network scenarios in the presence of background traffic. In the downlink case it was observed that when the QoS aware MPEG-4 video delivery algorithm was not implemented, for all twelve video clips there was an average ~5% frame loss for all three frame types. This percentage of frame loss translated into ~10% -18% loss in bandwidth. However, when the ReTx mechanism was applied the frame loss rate reduced to the target $\leq$ 1% level which means that more frames could be delivered successfully and consequently the net savings in bandwidth were observed in the range of ~9% -17%. In the uplink scenario, it was demonstrated that the network's video capacity is 2 streams. Afterwards the maximum tolerable background loads for 1 and 2 video streams were obtained which were 2.4 and 0.5 Mbps respectively. It was noticed that to obtain zero buffer occupancy after employing frame ReTx, GOPTs in the region of 20% to 60% were required for different video clips.

5.4 Limitations of the Proposed QoS Aware MPEG-4 Video Delivery Algorithm and Its Implementation

It can be noted that the impact of different levels of GOPT on the QoD is investigated here but the impact of various GOPT levels on the QoE is beyond the scope of this work. It may be further investigated to establish which type of clip is capable of tolerating a greater level of GOPT from a QoE perspective. Also due to time constraints and complexities involved in implementation, transmission loss due to noise and interference, and line rate adaptation have not been considered here. Losses due to MAC collisions tend to be greater than transmission losses and hence a lossless channel is assumed. If the line rate changes due to the line rate adaptation mechanism reacting to changes in the channel conditions then the system capacity and the number of frame ReTx required would also change. If the line rate drops to 5.5 Mbps from 11 Mbps then it would take twice as long to transmit the same amount of data. Consequently, the average service rate would decrease resulting in a higher probability of buffer overflow. Hence more GOPT would be required to reduce the probability of buffer overflow. When transmission loss due to noise and interference is accounted for, it would cause the frame ReTx and level of GOPT to increase. Hence it is recognized that these are omissions from this work which would have impact on the performance of the QoS aware MPEG-4 video delivery algorithm. It is suggested that further work in this area should be undertaken to determine the impact of GOPT on the QoE for different encoding types and determining the impact of line rate adaptation on the performance of the proposed algorithm.

The algorithm was implemented to achieve a maximum 1% loss rate for a video stream by setting a target 1% loss rate for each of the three frames types separately. The algorithm implementation can be realized more efficiently by prioritising the re-transmissions in accordance with the relative priority of the frames ($I > P > B$).

Twelve MPEG-4 video clips of 5 minutes duration each were used for extracting modelling parameters for the simulation programs. Clips of such short duration do not reflect the full characterisation of any movie (typical Hollywood movie duration is approximately 100-120 minutes). Also, the audio content associated with the video was not considered during implementation of the QoS aware MPEG-4 video delivery algorithm. Hence if the algorithm could be evaluated for the full duration of any movie with integrated audio, the results (the level of GOPT and number of frame retransmissions required) would be more accurate than that was presented in this thesis. The algorithm should also be evaluated for different video encoding formats other than MPEG-4, e.g. H .263, H .264 etc.

The QoS aware MPEG-4 video delivery algorithm was implemented separately for each type of video clip. The performance of the algorithm may be further investigated for a mix of video clips, e.g. how would the algorithm perform if video clips with the largest and smallest frame sizes are transmitted on the same network.

No fragmentation of the video frames was considered while implementing the QoS aware MPEG-4 video delivery algorithm. According to the video frames analysis, the sizes of the *I, P* frames can significantly exceed these values for different clips. For example, the average maximum size of an AVATAR *I* frame is 9952 bytes. This is the maximum average value calculated for all frames. But the maximum frame sizes that can be transmitted over Ethernet and WLANs are 1500 and 2304 bytes respectively (this is the fragmentation threshold). In reality frames larger than the fragmentation threshold must be fragmented. Intelligent fragmentation size may help improve reliability in the presence of interference. In environments with severe interference, encouraging fragmentation by decreasing this threshold may improve the effective throughput. When single fragments are lost, only the lost fragment must be retransmitted. By definition, the lost fragment is shorter than the entire frame and thus takes a shorter amount of time to transmit. Setting this threshold is a fine balancing act. If it is decreased too much, the effective throughput falls because of the additional time required to acknowledge each fragment. Likewise, setting this parameter too high may decrease effective throughput by allowing large frames to be corrupted, thus increasing the retransmission load on the radio channel. The optimum level of fragmentation might be investigated in future which would provide the best performance in terms of throughput, loss rate for the streamed video etc. [158]

The QoS aware MPEG-4 video delivery algorithm was presented and verified for streamed video over IEEE 802.11b WLANs. In the future, the performance of the algorithm for real time video should be evaluated. Real time video has a stricter delay rate (150 ms - 400 ms) than streamed video (acceptable delay 10 sec.). So for real time video both delay and loss rate need to be considered for evaluating the performance of the algorithm.

Downlink video streaming is likely to be the most prevalent deployment scenario and consequently this scenario should receive the greater treatment compared to uplink if this QoS aware MPEG-4 video delivery algorithm were to implement in real life networks. But the proposed algorithm could not be implemented completely for the downlink scenario (ReTx was implemented, but GOPT was not) as there was very high buffer occupancy observed in the video queue due to the high number of video traffic sending stations (e.g. up to 17 stations for MZ clip.).

To implement the proposed QoS aware MPEG-4 video delivery algorithm in real networks, several important issues need to be considered -

a) There has to be some form of mechanism to separate a video stream into its constituent *I, P, B* frames at the server and to recombine the *I, P, B* frames to generate a video stream at the client in real time for the end user.

b) Synchronization between the *IPB* frames and the audio data is important as audio to video synchronization is a mandatory mechanism to be implemented in real-time multimedia applications. Any misalignment can negatively affect the QoE. The correspondence between audio and video frames is given by their timestamps. Specialized time functions need to be implemented in the QoS aware MPEG-4 video delivery algorithm to synchronize audio and video.

c) Transmission losses need to be considered in order to get more accurate results of loss rate which would in turn improve the performance of the proposed QoS aware MPEG-4 video delivery algorithm.

5.5 Summary of the Chapter

Incoming video content data, their packet sizes, packet rates, the level of station contention etc. have a huge impact on the performance of IEEE 802.11b networks for large volumes of video traffic data. The capacity of the WLAN is proportional to the size of the packets and inversely related to the contention for access. As there is not much bandwidth available in IEEE 802.11b networks, the streamed video quality starts degrading when number of streams increases on the medium. The quality of the received video is subject to degradation arising from packet loss, delay and jitter.

In this chapter a study of the nature of the interaction between IEEE 802.11b WLAN and streamed video has been described in terms of a novel QoS aware MPEG-4 video delivery algorithm. In summary, the algorithm is an adaptive GOP truncation (GOPT) scheme combined with Re-transmission (ReTx) of dropped frames which replaces uncontrolled packet loss with controlled packet loss resulting in quality degradation in a more graceful manner. The goal is to optimise the delivery of video frames on a WLAN network that is being subject to bandwidth constraints.

By exploiting the interdependency of the constituent MPEG-4 video frames ($I > P > B$, $P_1 > P_2 > P_3 > P_4$, $B_1 > B_2 > B_3 > B_4$ *…etc)*, the GOPT mechanism may be implemented to discard the less important video frames in triplets and frame ReTx may be used to retransmit the lost frames in order to reduce the effect of degradation in video quality when the network resources are scarce. In other words, dropping off incoming packets in order of their importance is taking place to address the bandwidth constraints.

Twelve different video clips of six different genres were used to validate the QoS aware MPEG-4 video delivery algorithm. Each of the twelve video clips was 300 seconds long and prepared to follow the MPEG-4 standard (ASP profile) by encoding them using the FFMPEG software application. At first they were analysed to extract various important modelling parameters regarding their constituent *I, P, B* video frames (frame number, sizes etc.). Computer programs were written in the *C* programming language for simulating the uplink and downlink topologies separately in the IEEE 802.11b wireless network.

In the downlink case both video and background traffic were simulated to drive the network into saturation to evaluate the performance of the streamed video. It was observed that when the QoS aware MPEG-4 video delivery algorithm was not implemented, for all twelve video clips there was an average ~5% frame loss for all three frame types. This percentage of frame loss translated into ~10% -18% loss in bandwidth. However, when the ReTx mechanism was applied the frame loss rate reduced to the target $\leq$ 1% level which means that more frames could be delivered successfully and consequently the net savings in bandwidth were observed in the range of ~9% -17%. Hence, through computer simulation it was shown that the algorithm replaces uncontrolled frame loss with a more graceful prioritized frame loss.

In the uplink scenario, several stations were contending to access the medium before transmitting data. The “Video Capacity” of the system was established by setting zero buffer occupancy targets for the video stations’ buffers. It was demonstrated that the network’s video capacity is 2 streams. Afterwards the maximum tolerable background loads for 1 and 2 video streams were obtained. To address the unacceptable frames loss rates, frame ReTx was employed to improve reliability. When frame ReTx was used, the resulting buffer occupancy was greater than zero. So to compensate for this (i.e. to bring the average buffer occupancies back to zero) GOPT was applied. It was noticed that the level of contention present in the medium and video content type were attributed to for different levels of GOPT required to obtain the target loss rate and zero video buffer occupancies. The levels of GOPT and ReTx were found to be related to the frame sizes, e.g. video clips with smaller frames sizes required higher level of GOPT. The more truncation that takes place the less bandwidth was required, i.e. the video stream would be able to tolerate more background traffic. The strategy of completely eliminating buffer overflow (i.e. a zero probability of buffer overflow) is overly conservative and hence the level of GOPT was found to be quite high. If a finite probability of buffer overflow were to be adopted the level of GOPT would be lower and would vary depending upon the nature of the video clip. In this work an overly conservative approach was adopted. It is evident that the proposed model operates on the buffer occupancy information and thus indirectly on the frame service and arrival rates in the buffer.

Hence with the algorithm employed, the loss is more controlled and the reduction in the quality is also gradual as opposed to a potentially catastrophic step drop in video quality. In summary, the results presented in this chapter demonstrate that the QoS aware MPEG-4 video delivery algorithm improves QoD for streaming video under the prevailing network conditions.

The proposed (and subsequently validated) algorithm is based upon the experimental results of appendix B where it was shown that the MPEG-4 video QoD over WLANs could be improved by exploiting the *IPB* frame based nature of MPEG-4 video. To the best of our knowledge this is the first time an algorithm involving streamed MPEG-4 video over IEEE 802.11 WLANs has been proposed and validated for both uplink and downlink network scenarios using the two mechanisms– ReTx and GOPT by exploiting the frame hierarchy information of the MPEG-4 videos.

Chapter 6 CONCLUSION AND FUTURE WORK

Delivering real-time services such as video along with best effort data traffic over WLANs requires differentiated and prioritized traffic mechanisms. The performance of video services is quite sensitive to packet loss and delay or other causes which may lead to screen freeze and audio quality distortion [159] and consequently the viewers experience would be unsatisfactory. To design and dimension a video streaming over WLAN system, some important issues need to be effectively dealt with so as to guarantee the reliability of the network. From a network engineer's perspective, bandwidth and QoS (which includes delay, loss, received bit rate etc.) are among the most important issues to be managed.

The bursty and time-varying nature of video and the particular characteristics of the wireless network make video streaming over WLANs quite demanding. There are many forms and types of encoded video e.g. MPEG-1, MPEG-2, MPEG-4, H.264, High Definition etc. They differ widely in terms of their output characteristics, i.e. different frame rate, resolution, bit-rate etc. WLANs exhibit undesirable characteristics like time-varying bandwidth, higher delay, jitter, and losses compared to wired networks which make video streaming even more challenging. WLANs operate in unlicensed public bands hence interference from other devices operating in the same frequency bands, e.g. Bluetooth, cordless phones etc. are a reality. Interference can have a negative impact [160] on the performance of video streaming over wireless LANs.

6.1 Conclusions

A novel QoS aware MPEG-4 video delivery algorithm was proposed in chapter 4 and it was implemented and validated in chapter 5 that improves streaming MPEG-4 video QoD over WLANs by exploiting the *IPB* frame based hierarchical nature of MPEG-4 video. To provide the proof of concept for the algorithm a computer model was developed in *C* programming language for both uplink and downlink video traffic in the presence of CBR background traffic. The features of IEEE 802.11b and MPEG-4 videos were incorporated in the computer model. Modelling parameters were extracted from twelve different 5 minutes long MPEG-4 video clips (selected from six genres: CGI, Action, Animation, Sport, Documentary, and Talking Head). The proposed novel QoS aware MPEG-4 video delivery algorithm primarily deals with the performance aspect related to frame losses.

There are three ways in which video frames can be lost on a WLAN. These are – MAC collisions arising from contention for access, buffer overflow due to an insufficient availability of transmission opportunities to satisfy the incoming video frames and transmission losses due to noise and interference present in the medium. The QoS aware MPEG-4 video delivery algorithm advocates the combined use of failed frame retransmission (ReTx) and GOP truncation (GOPT) and seeks to eliminate potentially catastrophic buffer overflow and to minimize frames lost due to MAC collisions in a bandwidth efficient manner. It and is based on the measurement of the buffer occupancy metric. Transmission loss was not considered when implementing the algorithm. The ReTx mechanism facilitates the management of frames losses arising from collisions due to contention thereby increasing the QoD at the expense of higher probability of buffer overflow by reducing the effective average buffer service rate. The GOPT mechanism is employed to reduce the probability of buffer overflow by discarding less important GOP frame triplets at the expense of a reduced QoE at the receiver. Discarding GOP frame triplets must be employed in such a way that the scheme has the least impact on the received video stream at the client. Thus the proposed scheme implements an *Adaptive Video Streaming System* by determining the optimal manner to combine the ReTx and GOPT mechanisms under the prevailing network conditions to eliminate buffer overflow and minimise transmission losses. The QoS aware MPEG-4 video delivery algorithm aims to replace uncontrolled packet loss due to buffer overflow and MAC collisions by a controlled prioritized packet loss scheme that permits graceful degradation. It achieves the ITU-T target specified for loss rate of streamed video transmission. The algorithm is generic in nature although it is proposed for IEEE 802.11b WLANs only. It would work with all types of IEEE 802.11 based WLANs (e.g. b/g/a/n) as it is concerned with buffer occupancy. Also, it is applicable not only to MPEG-4 format but also to a wide range of other video transmission formats (e.g. H.263/.264) which can be segregated into their constituent *IPB* frames.

For both uplink and downlink network scenarios the QoS aware MPEG-4 video delivery algorithm has been shown to provide a significant improvement in the QoS performance for video streaming applications in the presence of background traffic through extensive simulations. The main findings from the simulations carried out to validate the operation of the proposed algorithm are summarised below. The first four points are for uplink case and the remaining one is for downlink case.

Simulation results demonstrate that the probability of buffer overflow and the capacity of the network are essentially independent of the content type of the video streams. Contention depends on the number of MPEG-4 video streams competing for access to the medium. Capacity (in terms of bandwidth) may be defined as the maximum load that can be transmitted on the network before saturation and it decreases as the number of streams increases. The capacity of the system was found to be inversely related to the number of streams contending for access.

- In the absence of background traffic it was found out that the system has a Video Capacity of 2 MPEG-4 streams for all content types where the Video Capacity is defined by the maximum number of video streams that can be simultaneously accommodated without incurring buffer overflow.

- When CBR background traffic was introduced, buffer occupancy for the video streams was calculated at different levels of background traffic for different video contents. In was demonstrated that the maximum background traffic that can be tolerated before the video streams fail varied for different video contents. The maximum tolerable background loads for 1 and 2 video streams were obtained which were 2.4 and 0.5 Mbps respectively. It was found that 2

AVATAR and 2 MARK ZUCKERBERG video streams separately could tolerate a maximum ~550 kbps and ~500 kbps of background load (comprising 1500 byte size packets) respectively. The video clips with larger average packets showed comparatively greater resilience than the video clips with smaller packet sizes. The reason is that the smaller the average video packet size the greater the bandwidth loss to stations with greater average packet sizes. So system capacity in terms of throughput differs from one content type to another but video capacity in terms of number of streams is the same for all content types.

- Different levels of GOPT were required for different video contents. It was noticed that to obtain zero buffer occupancy after employing frame ReTx, GOPTs in the region of 20% to 60% were required for different video clips. The levels of GOPT and ReTx were found to be related to the frame sizes, e.g. video clips with smaller frames sizes required higher level of GOPT. The more GOP truncation that takes place the less bandwidth was required, i.e. the video stream would be able to tolerate more background traffic. However, the greater the impact on the QoE. The strategy of eliminating buffer overflow altogether (i.e. a zero probability of buffer overflow) is rather conservative and hence the level of GOPT was found to be quite high.

- In the downlink case when the QoS aware MPEG-4 video delivery algorithm was not implemented, for all twelve video clips there was on average ~5%

frame loss for all three frame types (*IPB)* which translated into a ~10 to ~18% loss in bandwidth. When the algorithm was applied the frame loss was reduced below the target of < 1%. Hence the net savings in bandwidth were found to be in the range of ~10% as savings were realised through avoiding transmitting undecodable video frames.

Through extensive simulation it was shown by the implementation of the novel QoS aware MPEG-4 video delivery algorithm that there was a gradual decrease as opposed to sudden drop in video quality. A significant improvement in the QoS performance was observed for video streaming applications for both uplink and downlink network scenarios in the presence of background traffic. The proposed efficient QoS aware MPEG-4 video delivery algorithm eliminates unpredictability and provides the most favourable operating conditions for the video streams under the prevailing network conditions. The proposed algorithm is also generic in nature, i.e. as it is concerned with buffer occupancy; it can work with all types of IEEE 802.11 based WLANs (e.g. b/g/a/n). To the best of our knowledge, this is the first time where a solution has been proposed and validated for enhancing the quality of streamed video over WLANs by breaking up the MPEG-4 video into its constitute frames and then by combining the ReTX and GOPT mechanisms to minimize frame losses and eliminate potentially catastrophic buffer overflow .

6.2 Future Work

Further research relevant to the experimental study presented may include the following topics:

- The IEEE 802.11b standard has a maximum physical data rate of 11 Mbps and is able of supporting MPEG-4 encoded standard-definition (SD) video, while IEEE 802.11a/g (54 Mbps) can carry high-definition (HD) video [109]. The performance of the new IEEE 802.11n networks which promises a throughput of over 100 Mbps can be evaluated for by streaming ultra high definition (UHD) video over this new standard.

- By providing prioritized access to the audio and video streams over best-effort data traffic through an appropriate tuning of the *AIFSN*, CW_{min}, CW_{max} settings in conjunction with the *TXOP Limit* parameter network performance can be evaluated. An interesting research topic would be tuning the IEEE 802.11e EDCA parameters for improved video quality output from the network, with and without background traffic with many stations contending.

- There are various streaming servers available (example of open source servers are Darwin Streaming Server (DSS) and Helix from Real. Performance of different streaming servers along with different types of video and audio streams can be evaluated. With the knowledge of appropriate server and multimedia content, a network engineer can design a WLAN system which will guarantee acceptable quality of service.

Possible Areas of Further Improvements for the QoS aware MPEG-4 video delivery algorithm:

- The algorithm was presented and verified for streamed video over IEEE 802.11b WLANs. In future the performance of the algorithm for real time video should be evaluated. Real time video has a stricter delay rate (150 ms - 400 ms) than streamed video (acceptable delay 10 sec.). So for real time video both delay and loss rate need to be considered for optimizing the algorithm.

- The target packet loss rate was 1% as described by the ITU-T [15,16] for streamed video. In this work, a 1% loss rate was adopted for all three different frame types (*IPB).* A future direction of this work might be to examine the optimum combination of the *IPB* loss rates to give a cumulative 1% loss rate for the video stream.

- Fragmentation of video frames was not considered. According to the video frames analysis, the sizes of the *I, P* frames were quite large for clips of different content. For example, the average maximum size of an AVATAR *I*-frame is 9952 bytes. This is the maximum average value calculated for all frames. The IEEE 802.11b standard defines a maximum frame size of 2304 bytes in the wireless networks. In reality, this limit would usually not be achieved as Ethernet (IEEE 802.3) has a maximum packet size limit of 1500 bytes. Hence, in practice the average AVATAR *I-* frames would be fragmented. However IEEE 802.11n might help regarding fragmentation as it defines improved fragmentation techniques.

- The majority of the video codecs do not implement a 15 frame GOP structure as recommended in the MPEG-4 standard in real life. Hence given that the GOPT mechanism can be applied generically to any arbitrary GOP structure, it is believed that the proposed new QoS aware MPEG-4 video delivery algorithm would also be beneficial in these other non standard video over WLAN systems. A further direction of the work would be to investigate the performance of the system for these non-standard GOPs.

- By applying admission control to the incoming video frames and tuning the four IEEE 802.11e buffer Access Categories and the four associated EDCA parameters (CW_{min}, CW_{max}, *AIFSN, TXOP*) the effectiveness of the QoS aware MPEG-4 video delivery algorithm could be improved. Admission control and the IEEE 802.11e EDCA mechanism could be used to manage the arrival rate and service rate respectively of the video streams. Tuning of IEEE 802.11e parameters may be employed to control the outcomes from contention (i.e. by prioritising winning access opportunities) and hence less contention could be achieved. It can increase the MAC service rate and hence reduce the buffer overflow. It can be used to increase the available bandwidth for video traffic and hence the video QoS can be enhanced thus negating the impact of other traffic in the network. The AP may be given higher priority than the stations by appropriately setting lower *AIFSN* and CW values.

- The QoS aware MPEG-4 video delivery algorithm is implemented for streaming videos which have a relatively large acceptable delay (~ 10 sec.) limit but stricter loss rate (~ 1%). For analysis, 300 second long clips were used after converting them to standard MPEG-4 videos. It would be useful if the algorithm could be implemented for 'on the fly' videos, i.e. for real time video traffic as it would give us the opportunity to do subjective video quality analysis on the client side. Real time videos have stricter delay limit compared to streaming video clips.

- As mentioned earlier, the QoS aware MPEG-4 video delivery algorithm and the IEEE 802.11b networks (both downlink and uplink) were implemented in computer programs written in *C* language under Windows XP Operating System. Collected data were analysed in detail using *C* and *Perl* codes. Other network simulation tools as Network Simulator (NS-3), Optimized Network Engineering Tools [161] (OPNET), OMNeT++ [162] etc. might also be used to implement the algorithm in LINUX and Windows environments to validate the results presented in this thesis.

- The performance of standard (ASP profile) MPEG-4 videos were analysed for IEEE 802.11b WLANs in this work. It would be beneficial to investigate the impact/relevancy of the QoS aware MPEG-4 video delivery algorithm for forthcoming Ultra High Definition (UHD) videos (7680 x 4320 pixels) and high speed Gbps WLANs.

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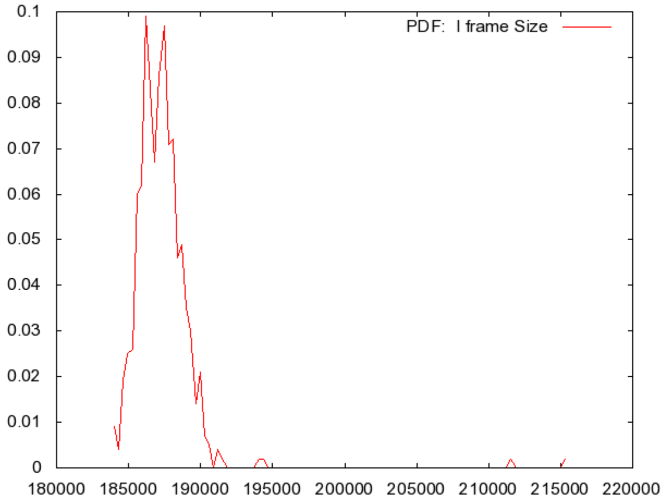
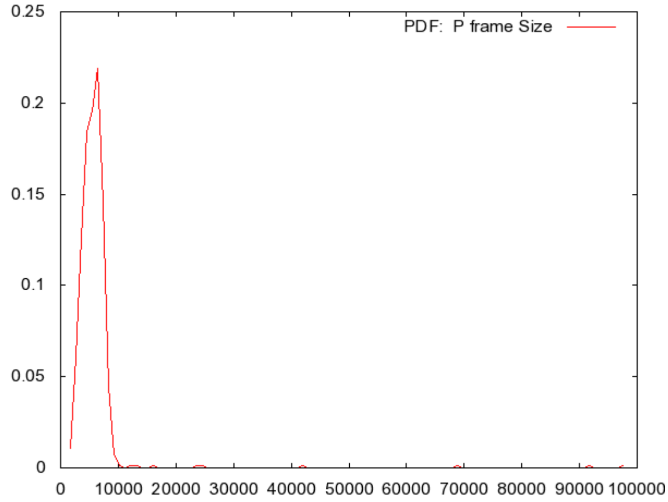
APPENDIX A

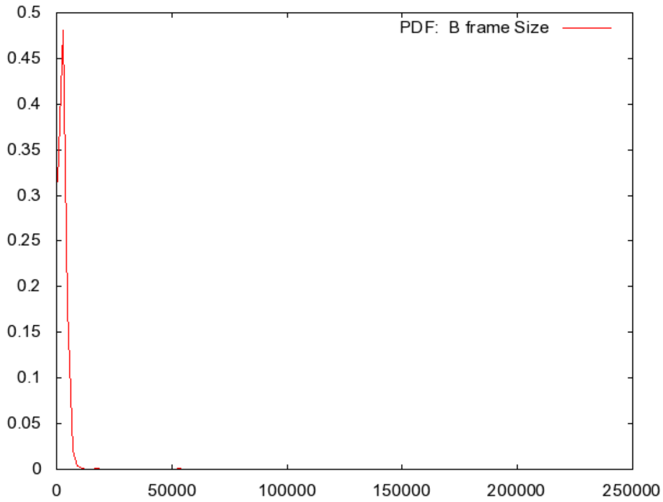
The PDFs of *I, P,* and *B* frames' and GOPs' are shown below for the 'AVATAR' video clip-



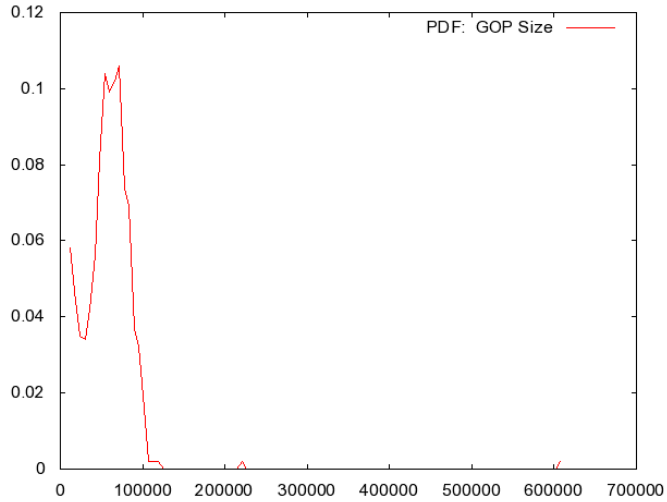
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



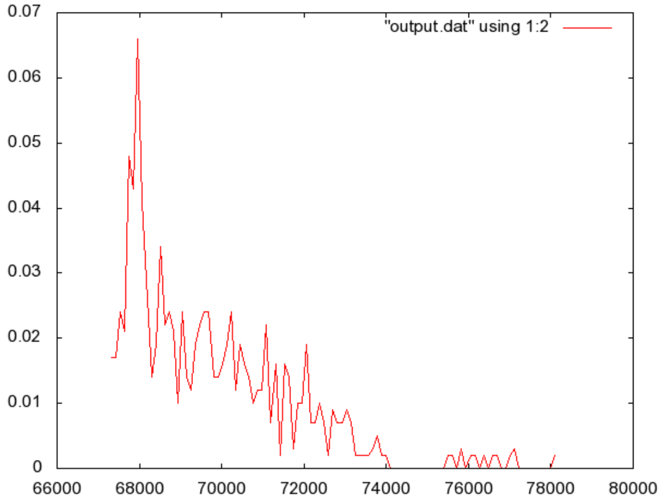
PDF of GOPs

2012

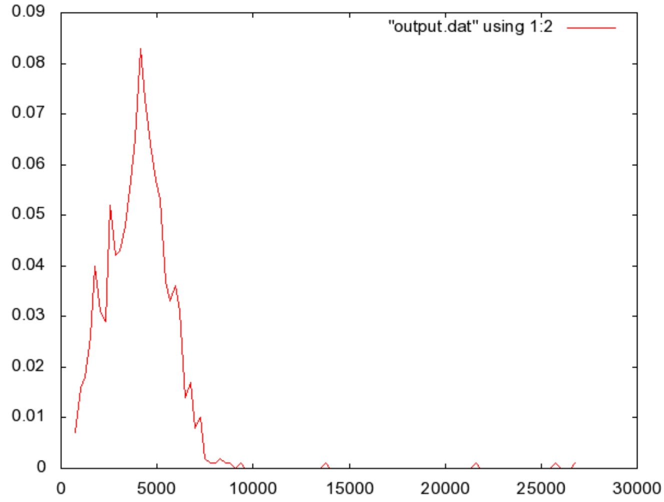
A ‘2012’ Movie Scene

Analysis of the ‘2012’ movie clip shows that the 300 second clip contains 580 *I* frames, 1921 *P* frames, and 5000 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 7310, 4167, and 2420 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 4.24, 8.01, and 12.10 Megabytes respectively. Among the 580 GOPs, the maximum, minimum and average GOP sizes were found to be 225.66, 7.15 and 41.98 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as 5459μs, 3173μs, and 1903 μs respectively.

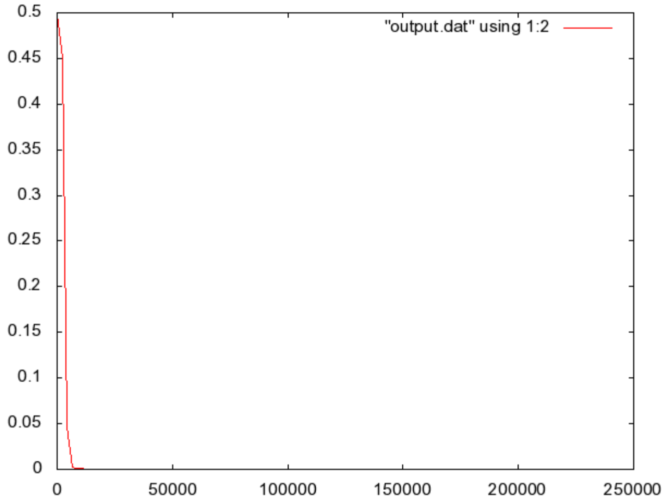
The PDFs of *I, P,* and *B* frames' and GOPs' are shown below for the '2012' video clip-



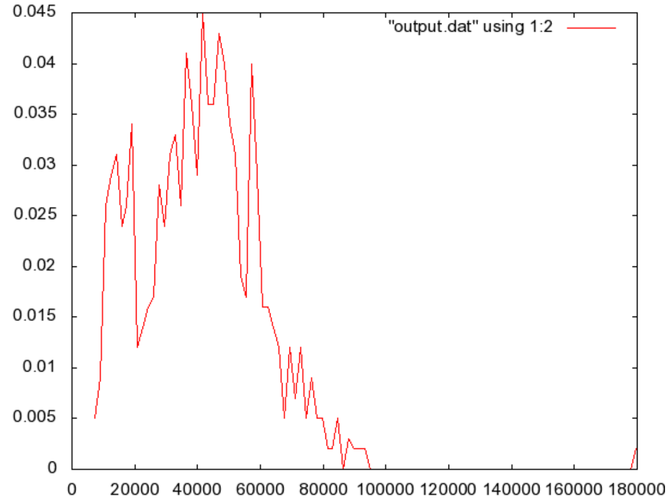
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



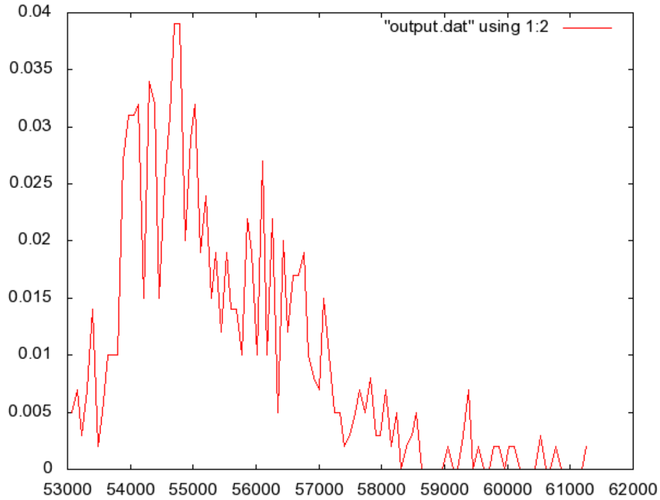
PDF of GOPs

DIE HARD 4

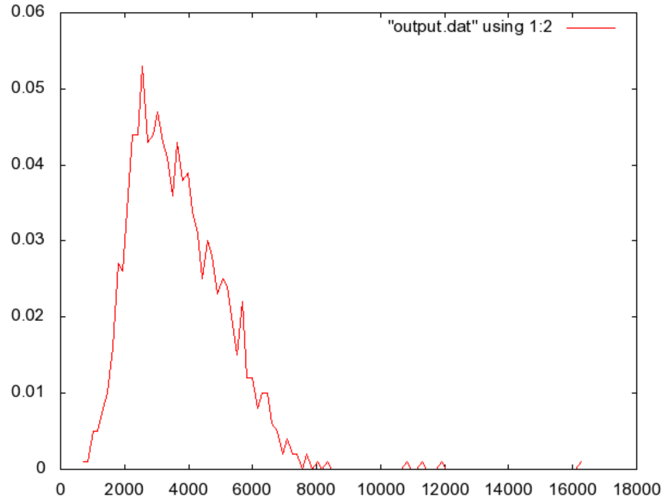
A Snapshot from the movie 'DIE HARD 4'

It has been found out that the 300 second long 'DIE HARD 4' movie clip contains 589 *I* frames, 1912 *P* frames, and 5000 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 7213, 3728, and 2318 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 4.25, 7.13, and 11.59 Megabytes respectively. Among the 589 GOPs, the maximum, minimum and average GOP sizes were found to be 247.77, 7.63 and 38.99 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as 5389μs, 2854μs, and 1828 μs respectively.

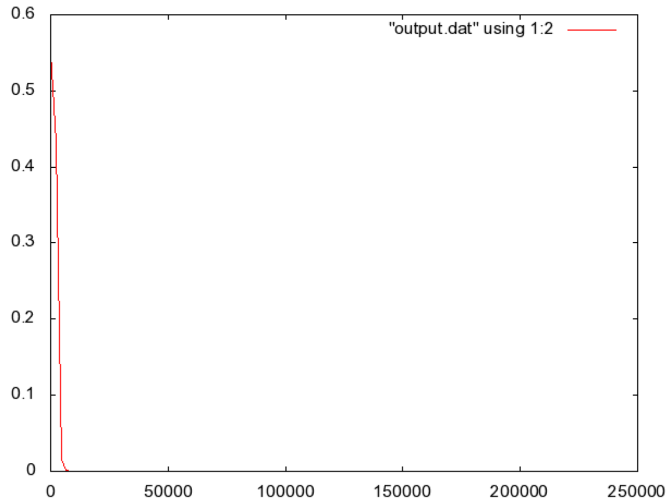
The PDFs of *I*, *P*, and *B* frames' and GOPs' are shown below for the 'DIE HARD 4' video clip-



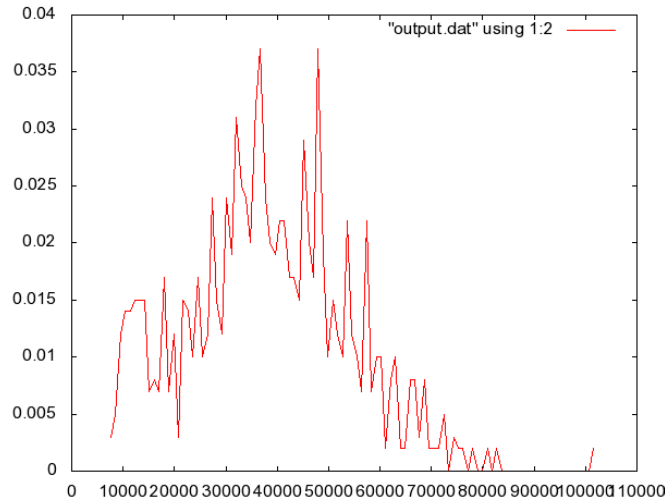
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



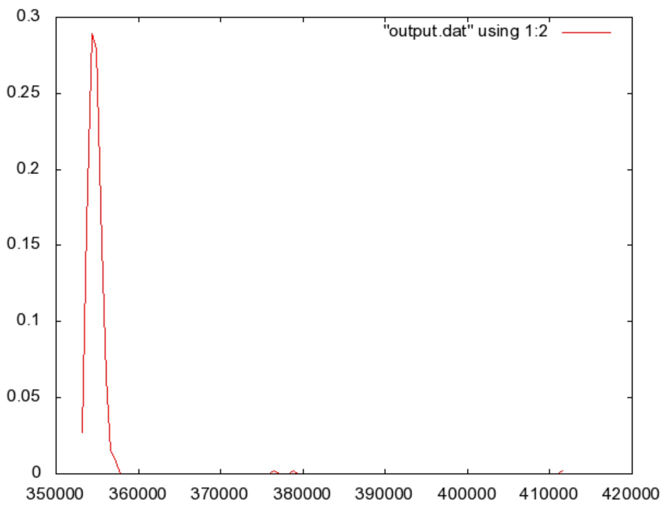
PDF of GOPs

KING ARTHUR

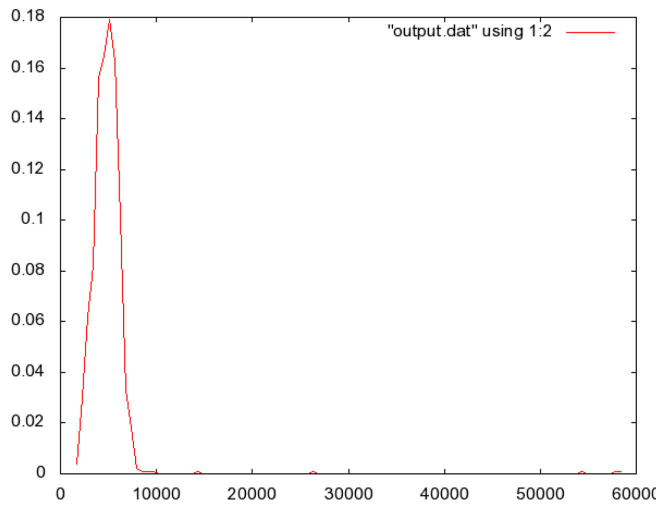
A Scene from 'KING ARTHUR' clip

Analysis of the 'KING ARTHUR' movie clip shows that the 300 second clip contains 596 *I* frames, 1907 *P* frames, and 5001 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 7427, 5158, and 3325 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 4.43, 9.84, and 16.63 Megabytes respectively. Among the 596 GOPs, the maximum, minimum and average GOP sizes were found to be 423.98, 10.21 and 51.83 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as 5544 μs, 3894 μs, and 2561 μs respectively.

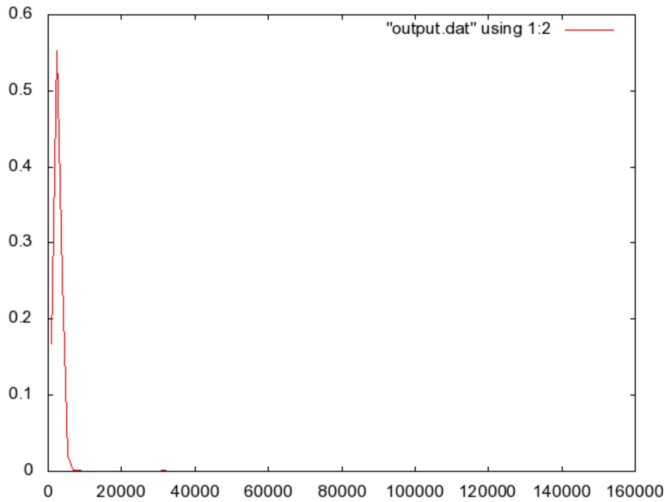
The PDFs of *I, P,* and *B* frames' and GOPs' are shown below for the 'KING ARTHUR' video clip-



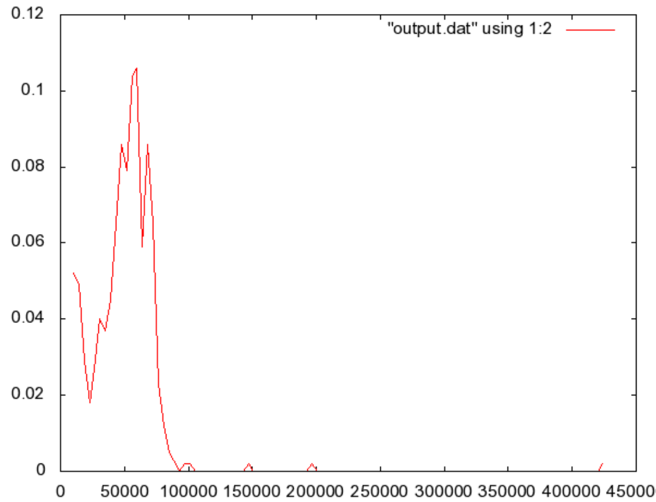
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



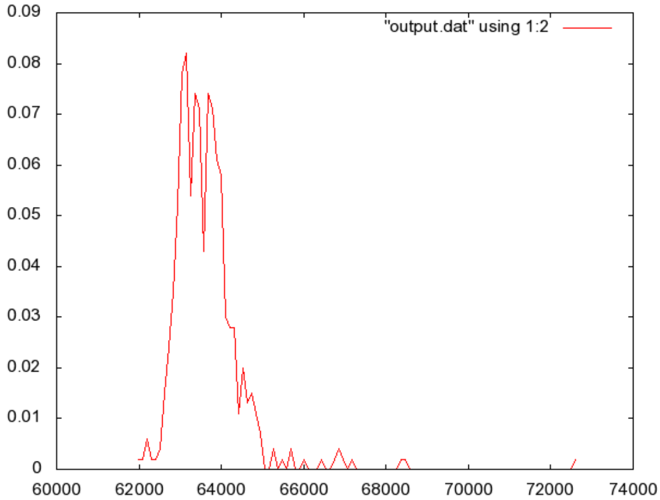
PDF of GOPs

LION KING 2

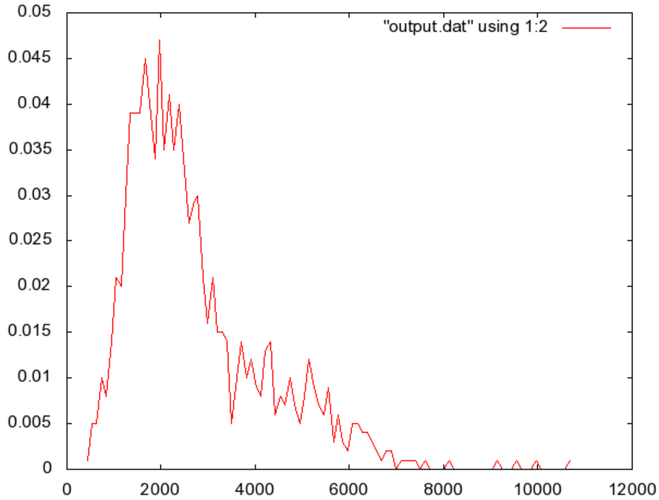
A Scene from the 'LION KING 2' Movie

After analyzing the 'LION KING 2' clip, it has been found out that the 300 second clip contains 538 *I* frames, 1964 *P* frames, and 4999 *B* frames. The average sizes of the *I, P,* and *B* frames have been calculated 6593, 2726, and 1364 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 3.55, 5.35, and 6.82 Megabytes respectively. Among the 538 GOPs, the maximum, minimum and average GOP sizes were found to be 74.56, 7.64 and 29.22 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as $4938\,\mu s$, $2125\,\mu s$, and $1135\,\mu s$ respectively.

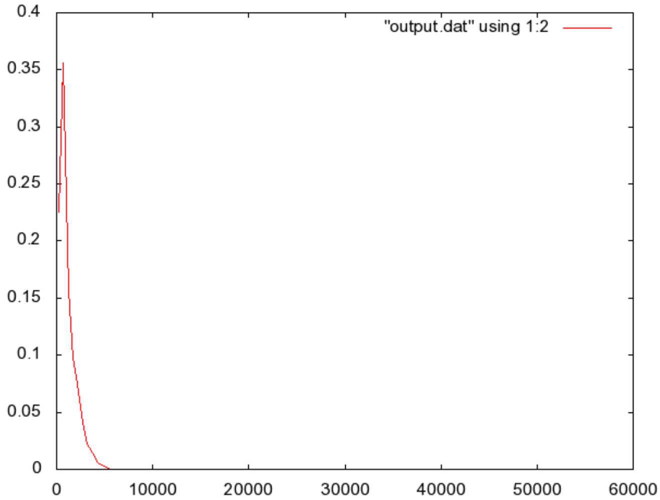
The PDFs of *I, P,* and *B* frames' and GOPs' are shown below for the 'LION KING 2' video clip-



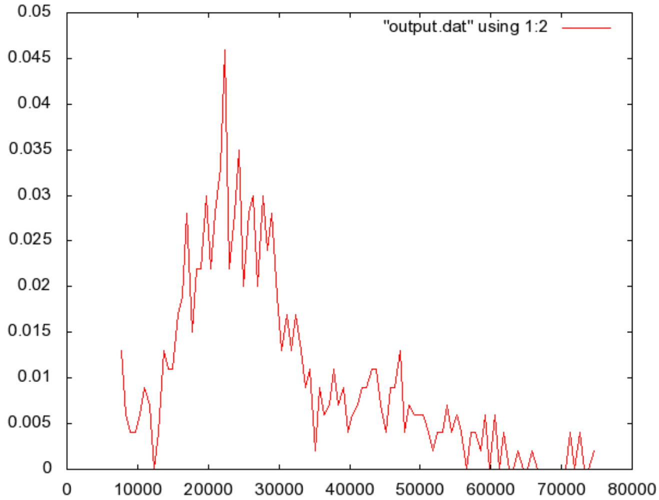
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



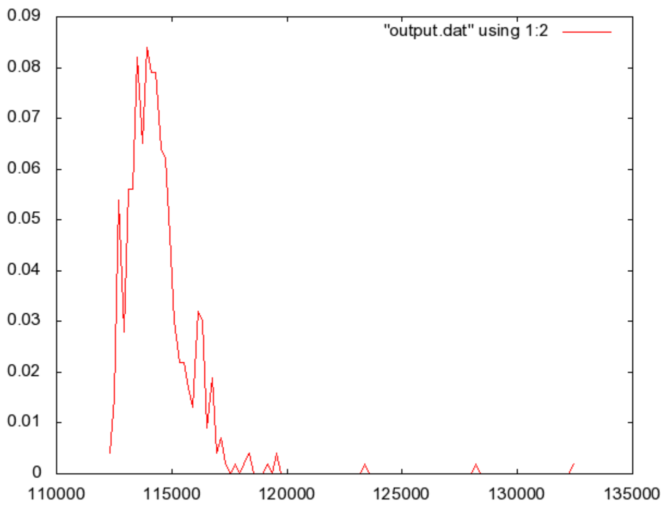
PDF of GOPs

ICE AGE 2

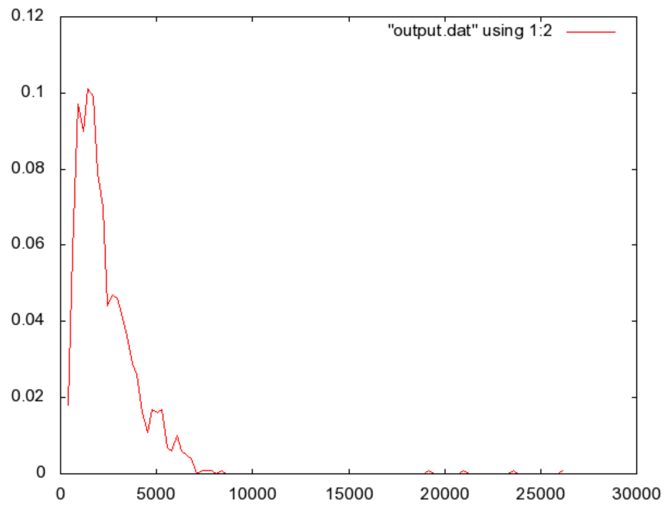
A Snapshot from the 'ICE AGE 2' movie

It has been found out that the 300 second long animated 'ICE AGE 2' movie clip contains 535 *I* frames, 1967 *P* frames, and 5000 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 6861, 2489, and 1374 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 3.67, 4.90, and 6.87 Megabytes respectively. Among the 535 GOPs, the maximum, minimum and average GOP sizes were found to be 151.45, 6.66 and 28.85 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as 5133 μs, 1953 μs, and 1142 μs respectively.

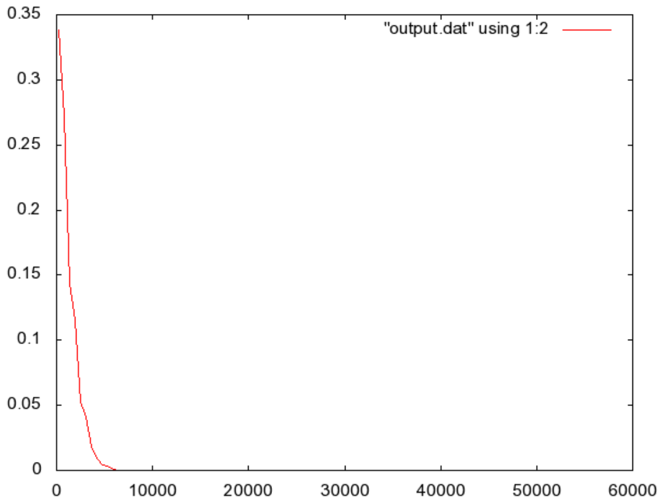
The PDFs of *I*, *P*, and *B* frames' and GOPs' are shown below for the 'ICE AGE 2' video clip-



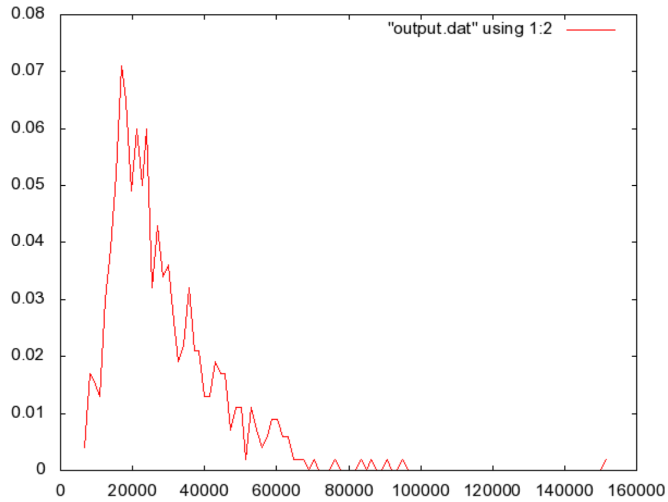
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



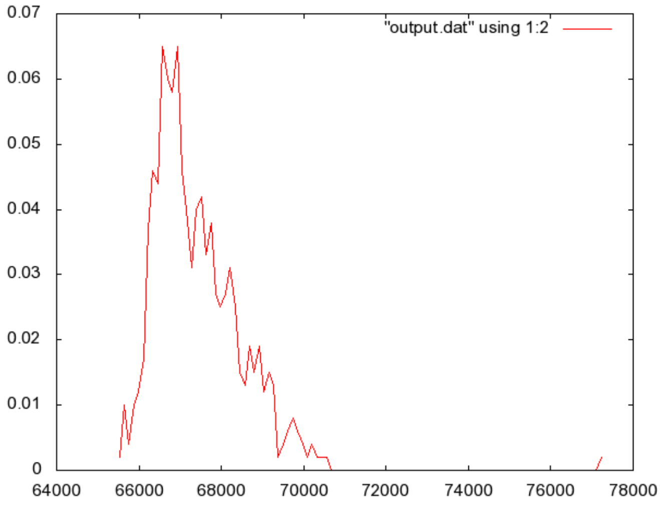
PDF of GOPs

RUGBY

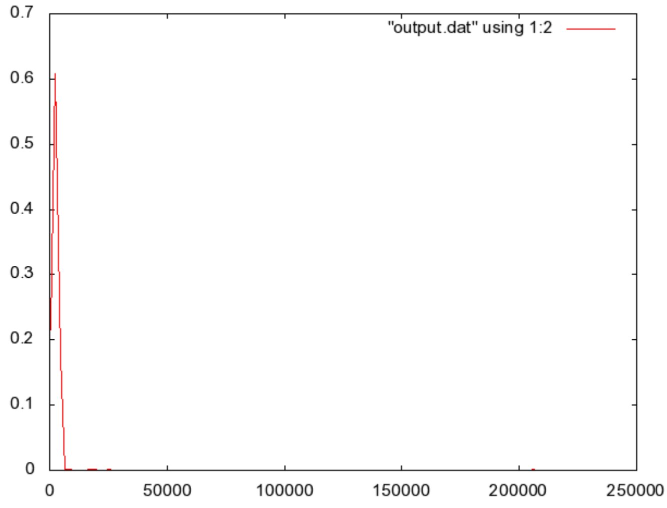
A Scene from the 'RUGBY' Clip

Analysis of the 'RUGBY' clip shows that the 300 second clip contains 520 *I* frames, 1982 *P* frames, and 5000 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 6607, 3603, and 1684 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 3.44, 7.14, and 8.42 Megabytes respectively. Among the 520 GOPs, the maximum, minimum and average GOP sizes were found to be 235.46, 7.04 and 36.54 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as $4948\mu s$, $2763\mu s$, and $1367\mu s$ respectively.

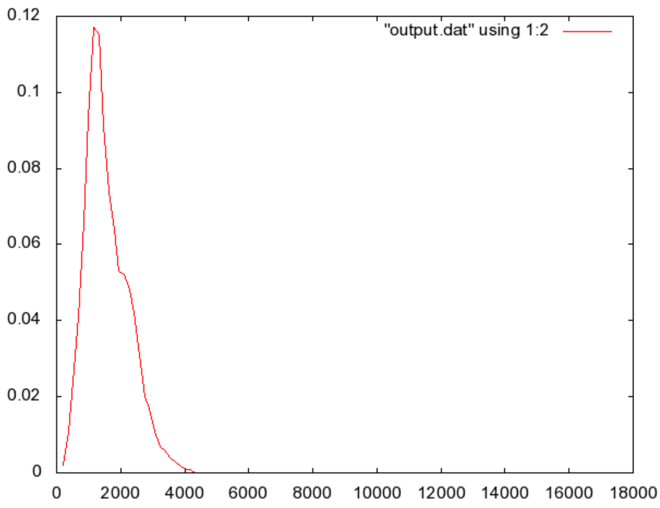
The PDFs of *I*, *P*, and *B* frames' and GOPs' are shown below for the 'RUGBY' video clip-



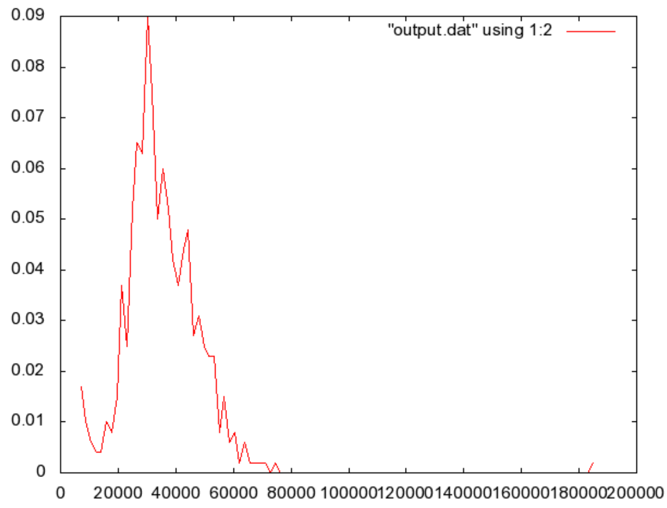
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



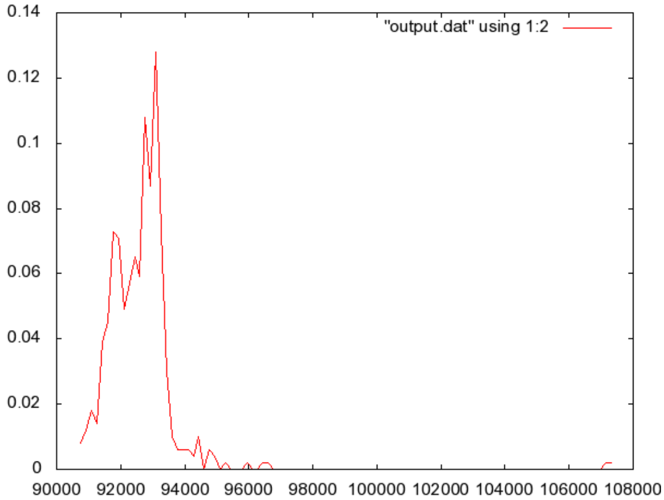
PDF of GOPs

FOOTBALL

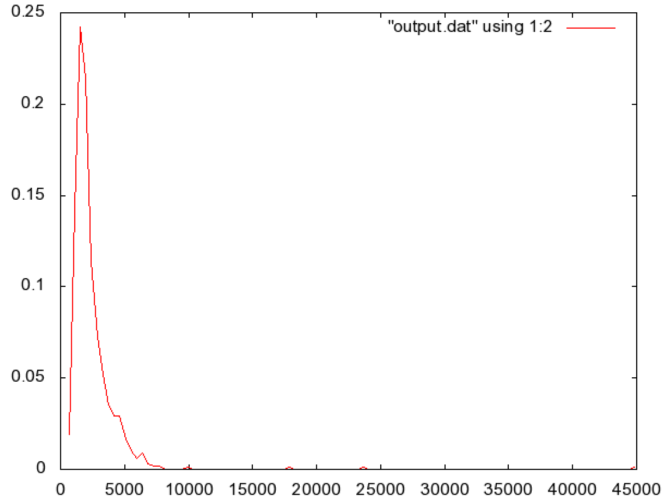
A Snapshot from the Brazil vs North Korea 2010 World Cup Match

After analyzing the ‘FOOTBALL’ clip, it has been found out that the 300 second clip contains 507 *I* frames, 1992 *P* frames, and 4992 *B* frames. The average sizes of the *I, P,* and *B* frames have been calculated 7994, 2550, and 1090 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 4.05, 5.08, and 5.94 Megabytes respectively. Among the 507 GOPs, the maximum, minimum and average GOP sizes were found to be 133.02, 7.43 and 29.73 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as 5957 μs, 1997 μs, and 1008 μs respectively.

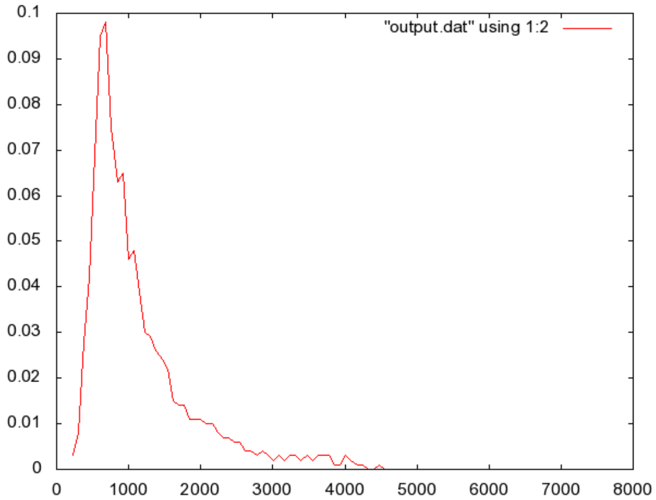
The PDFs of *I*, *P*, and *B* frames' and GOPs' are shown below for the 'FOOTBALL' video clip-



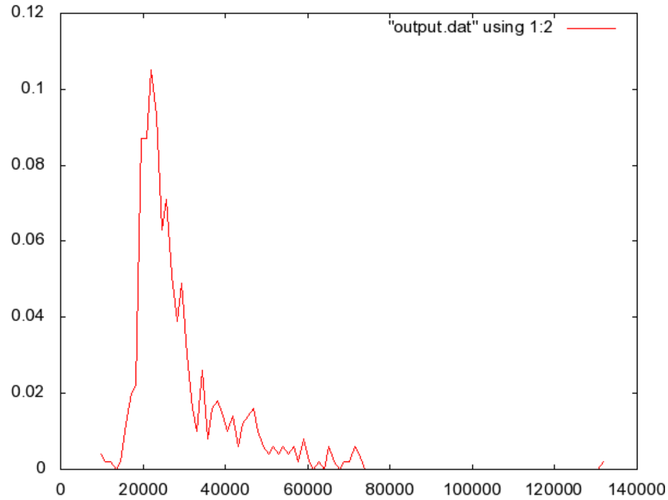
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



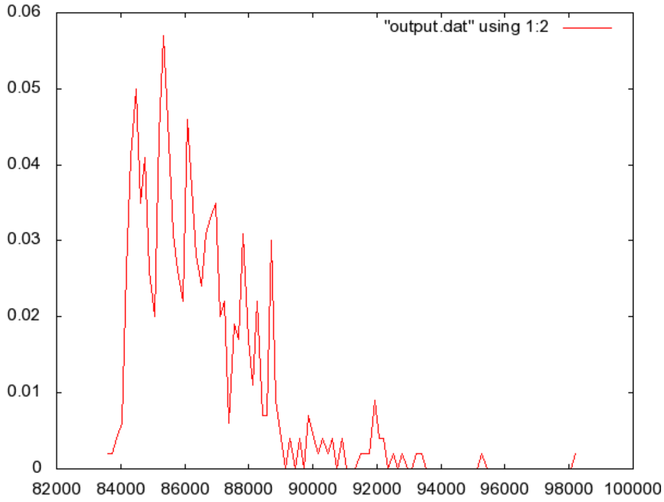
PDF of GOPs

BBC Documentary

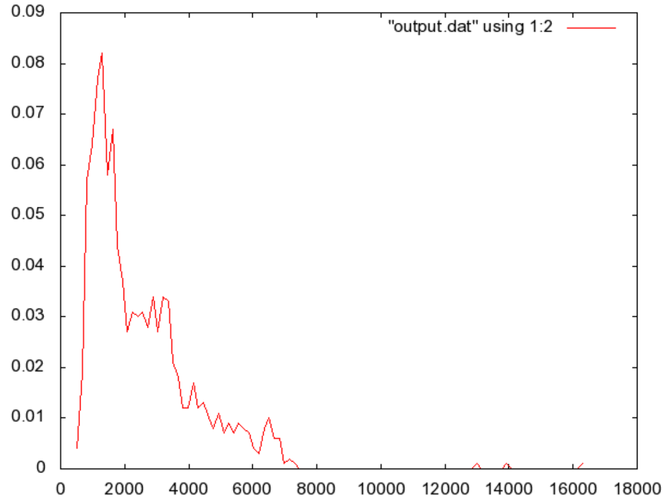
Snapshot Taken from the BBC Documentary -ICE WORLDS

It has been found out that the 300 second long documentary clip - 'ICE WORLDS' contains 540 *I* frames, 1962 *P* frames, and 5000 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 7605, 2566, and 1381 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 4.11, 5.03, and 6.91 Megabytes respectively. Among the 540 GOPs, the maximum, minimum and average GOP sizes were found to be 101.14, 4.7 and 29.72 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as $5674\mu s$, $2009\mu s$, and $1147\mu s$ respectively.

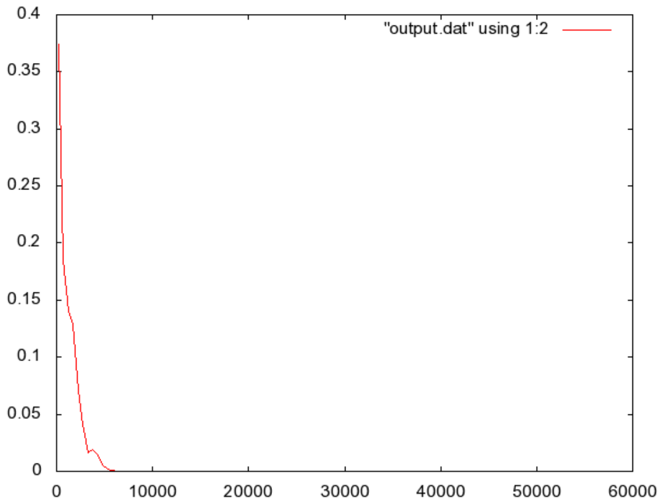
The PDFs of *I*, *P*, and *B* frames' and GOPs' are shown below for the 'ICE WORLDS ' video clip-



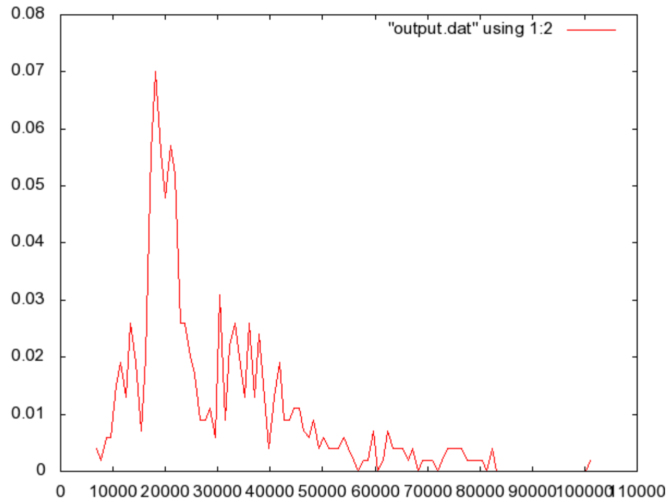
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



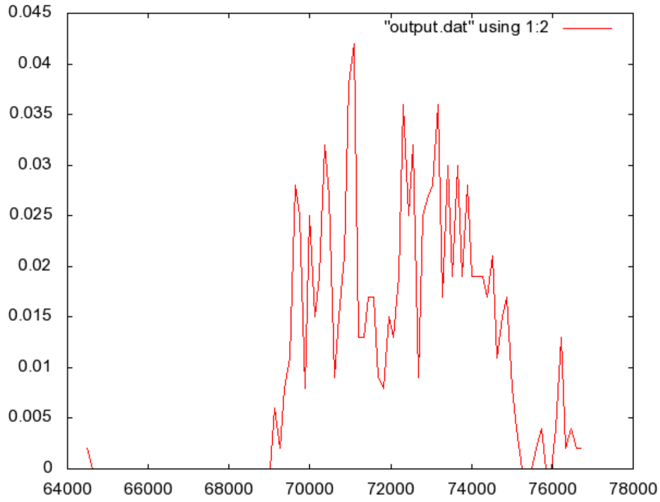
PDF of GOPs

Documentary: THE ANTARTICA CHALLENGE

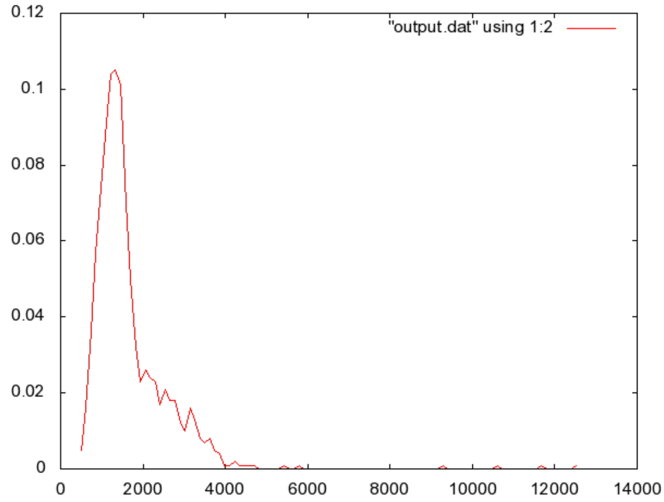
A Scene from the Documentary: THE ANTARTICA CHALLENGE-
A Global Warning

Analysis of the 'THE ANTARTICA CHALLENGE' documentary shows that the 300 second clip contains 528 *I* frames, 1974 *P* frames, and 5000 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 7904, 1704, and 849 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 4.17, 3.36, and 4.24 Megabytes respectively. Among the 528 GOPs, the maximum, minimum and average GOP sizes were found to be 850.63, 0.04 and 22.31 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as $5891\mu s$, $1382\mu s$, and $760\mu s$ respectively.

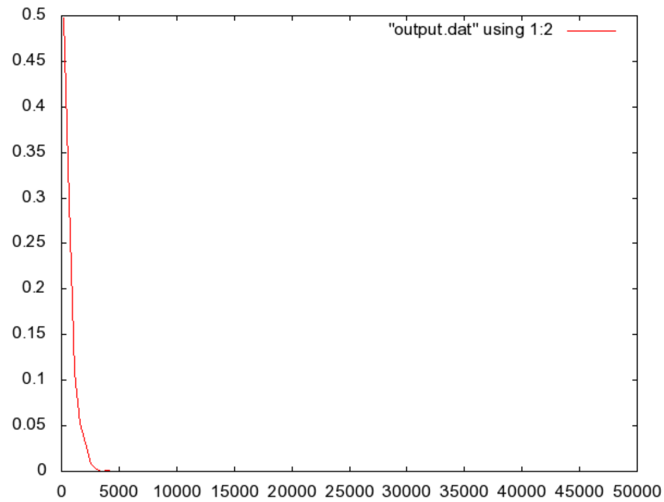
The PDFs of *I, P,* and *B* frames' and GOPs' are shown below for the 'THE ANTARTICA CHALLENGE' video clip-



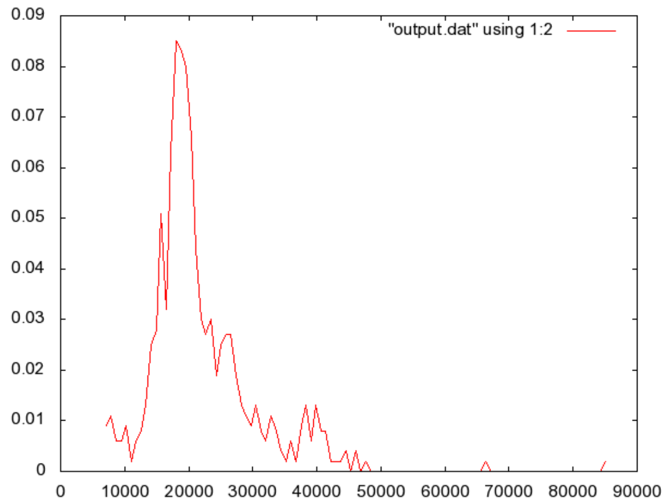
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



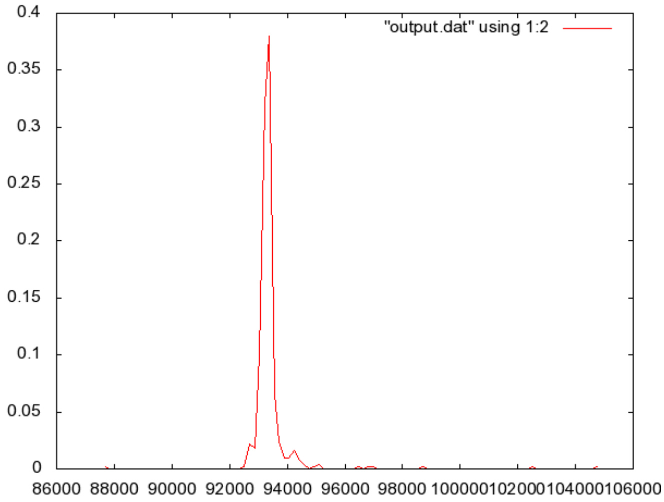
PDF of GOPs

Talking Head 1: Actor Matt Damon Interview

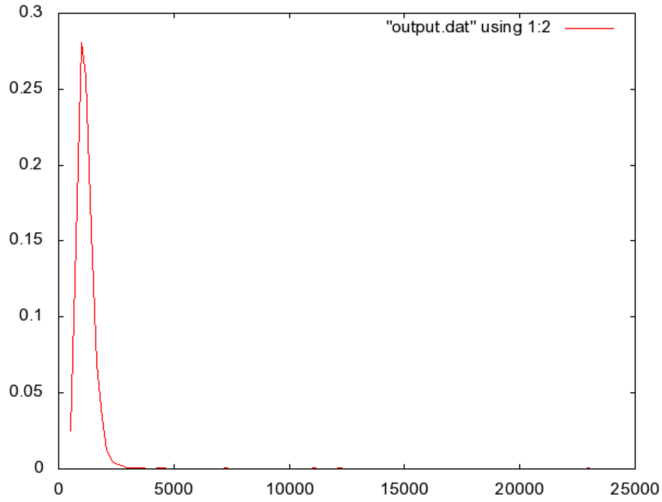
A Snapshot from the ‘Matt Damon Interview’

After analyzing the ‘Talking Head 1’ clip, it has been found out that the 300 second clip contains 498 *I* frames, 1987 *P* frames, and 4966 *B* frames. The average sizes of the *I, P,* and *B* frames have been calculated 7657, 1307, and 735 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 3.81, 2.60, and 3.65 Megabytes respectively. Among the 498 GOPs, the maximum, minimum and average GOP sizes were found to be 904.09, 16.05 and 20.20 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as $5711\mu s$, $1093\mu s$, and $677\mu s$ respectively.

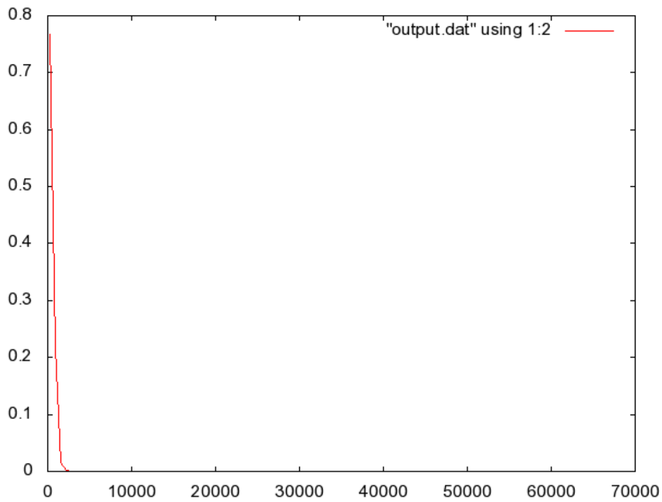
The PDFs of *I, P,* and *B* frames' and GOPs' are shown below for the 'Matt Damon Interview' test clip -



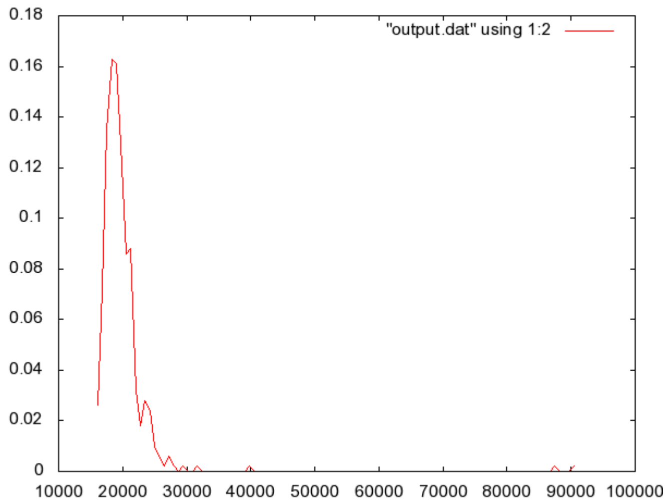
PDF of *I* Frames



PDF of *P* Frames



PDF of *B* Frames



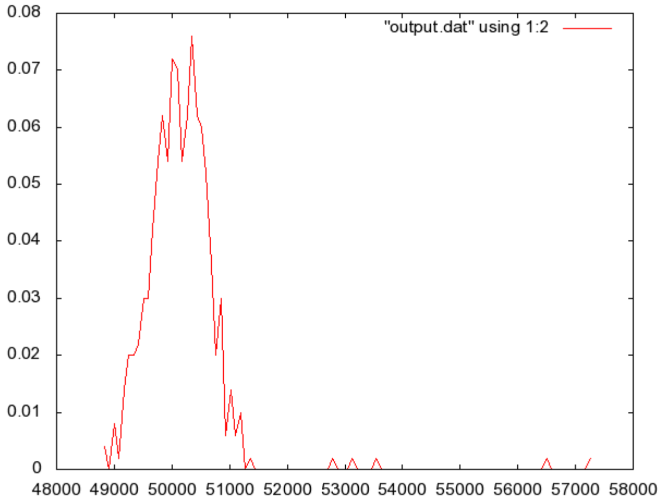
PDF of GOPs

Talking Head 2: Facebook Founder Mark Zuckerberg Interview

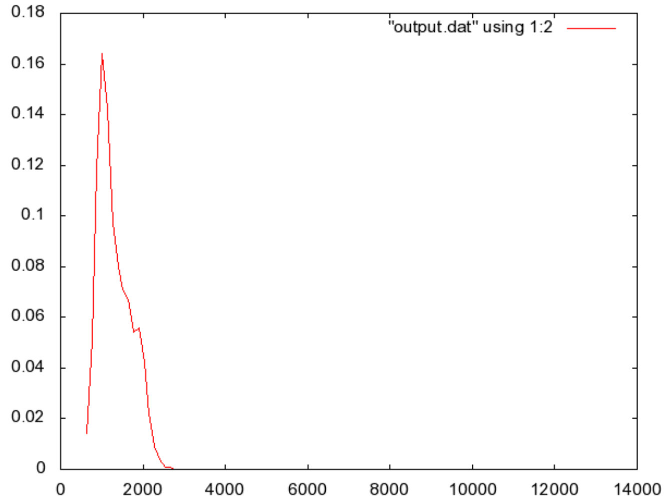
A Snapshot from the ‘Mark Zuckerberg Interview’

It has been found out that the 300 second long Mark Zuckerberg interview clip contains 502 *I* frames, 2002 *P* frames, and 5003 *B* frames. The average sizes of the *I, P,* and *B* frames have been found to be 7909, 1388, and 666 bytes respectively. So the total frame sizes of *I, P,* and *B* frames are 3.97, 2.78, and 3.33 Megabytes respectively. Among the 502 GOPs, the maximum, minimum and average GOP sizes were found to be 579.28, 14.21, and 20.08 kilobytes respectively. Frame transmission times (T_{MSDU}) for *I, P,* and *B* frames have been calculated as $5895\mu s$, $1152\mu s$, and $627\mu s$ respectively.

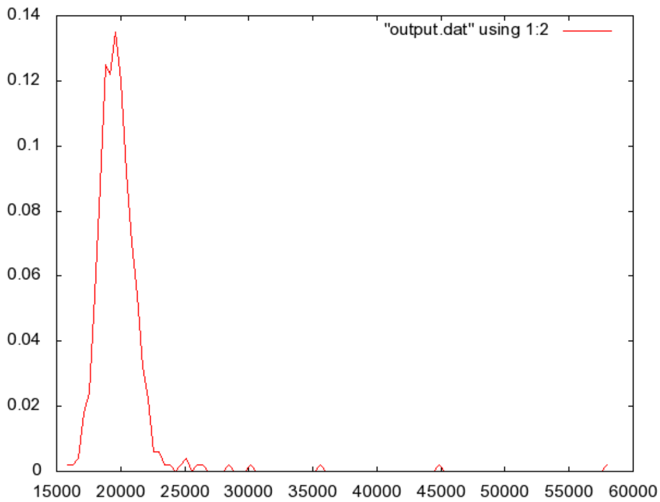
The PDFs of *I, P,* and *B* frames' and GOPs' are shown below for the 'Mark Zuckerberg Interview' video clip-



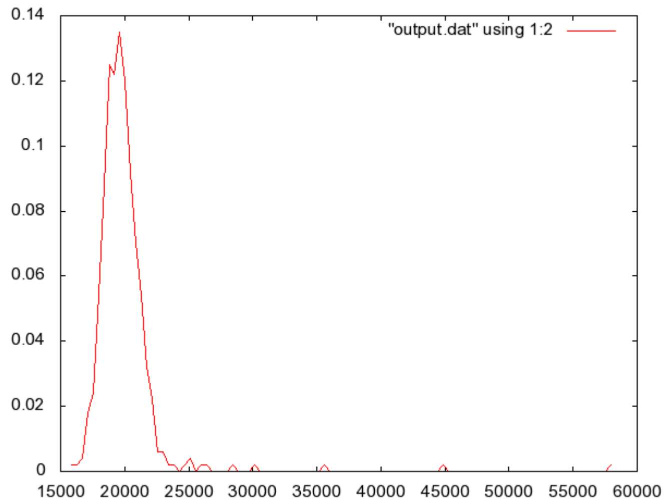
PDF of *I* Frames



PDF of *P* Frames

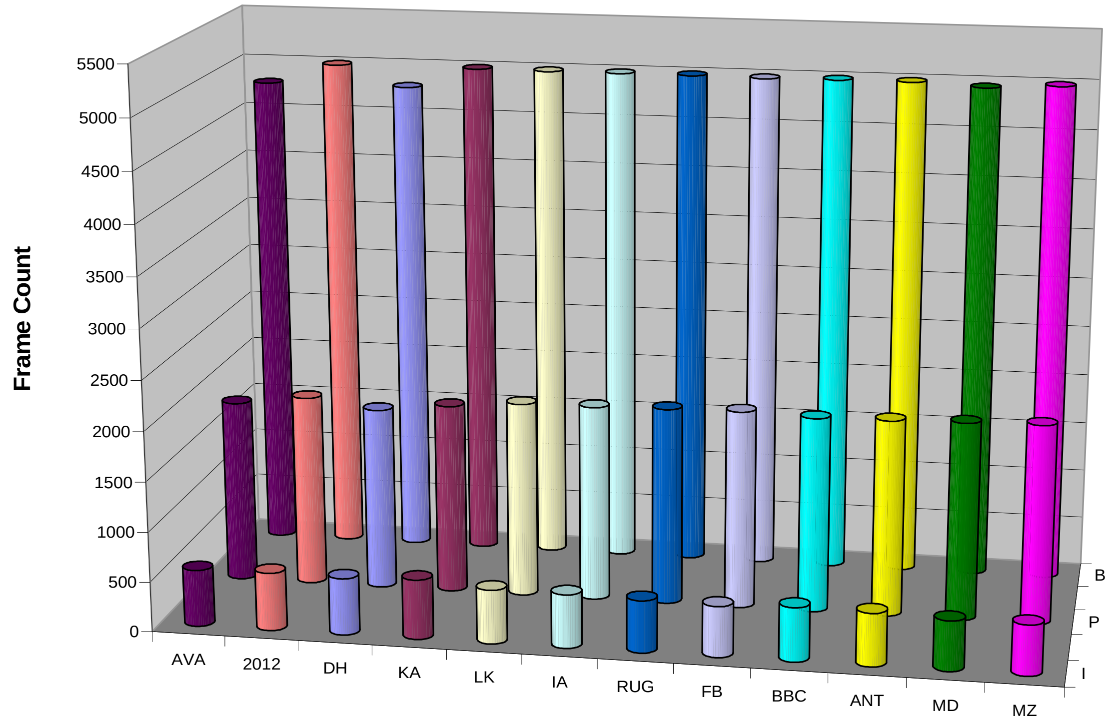


PDF of *B* Frames

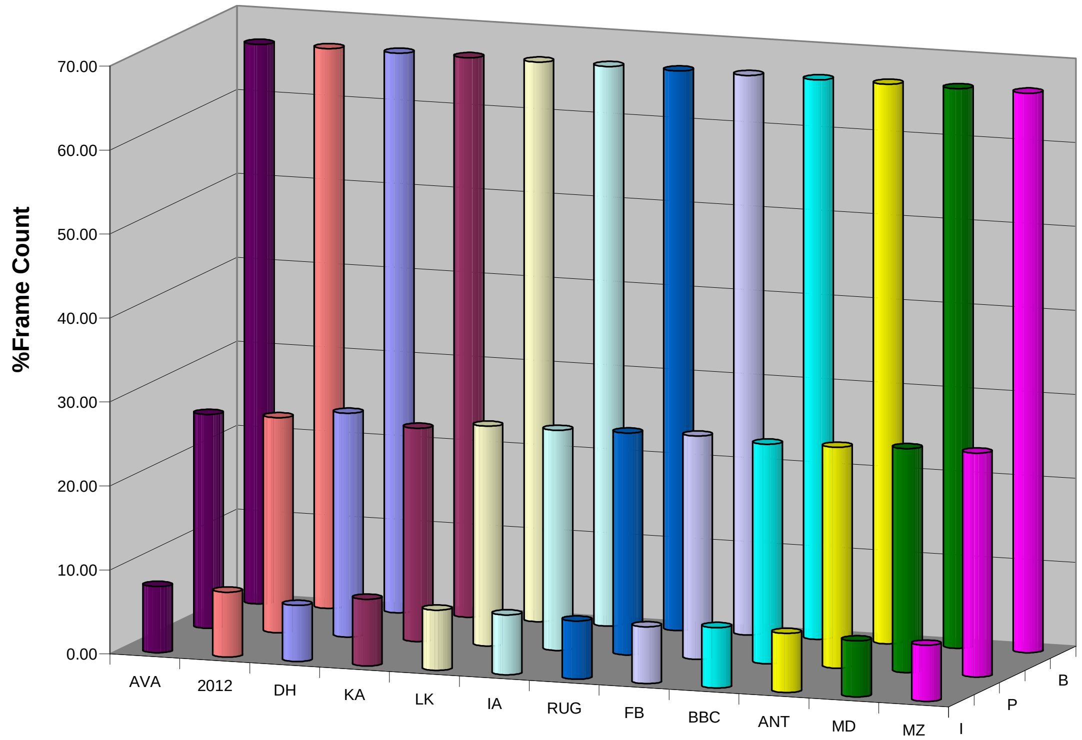


PDF of GOPs

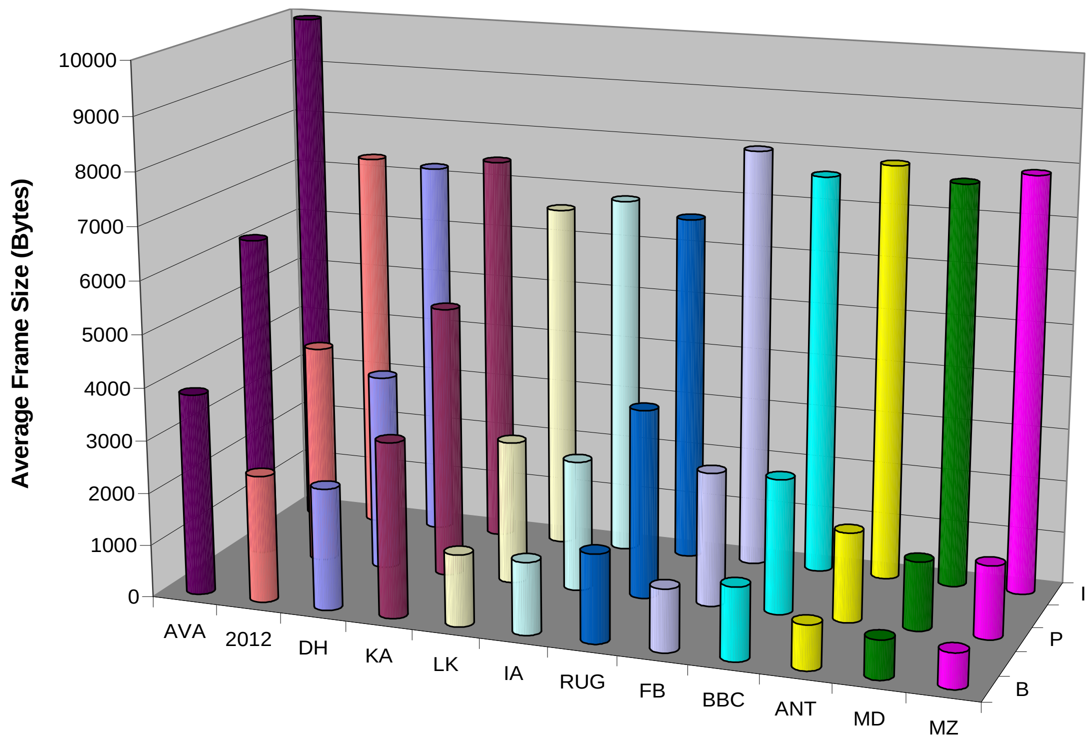
The following figures show different important parameters namely Frame Count, Average Frame Sizes, Total Frame Sizes, Medium Access Time for Different Frames, and GOP Sizes for all the 12 test video clips-



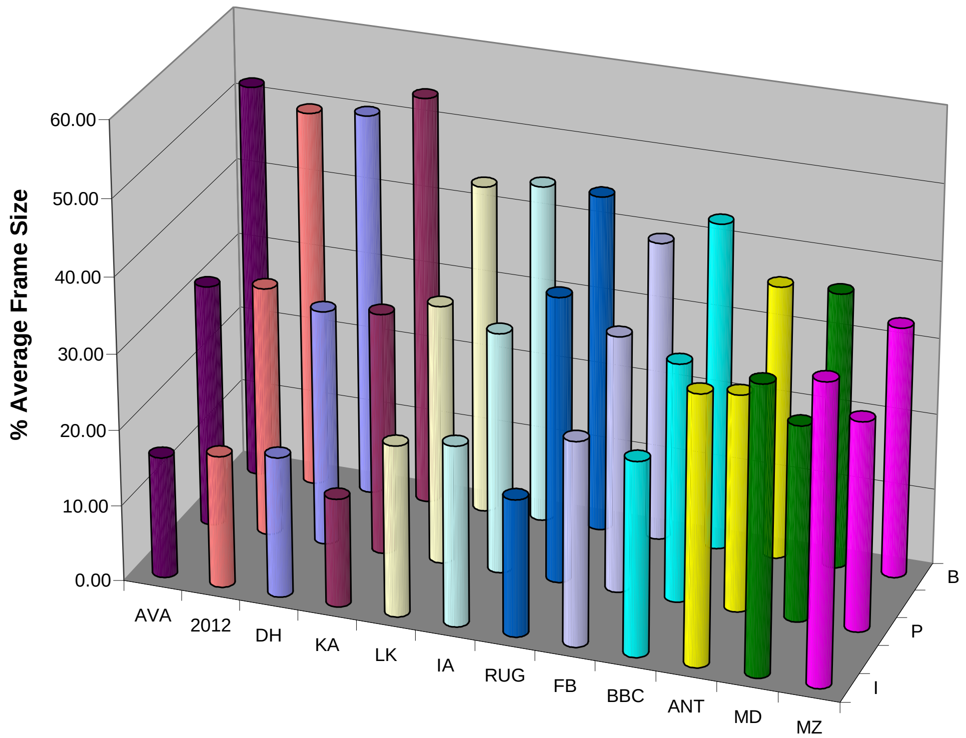
Frame Count of All the 12 Test Clips



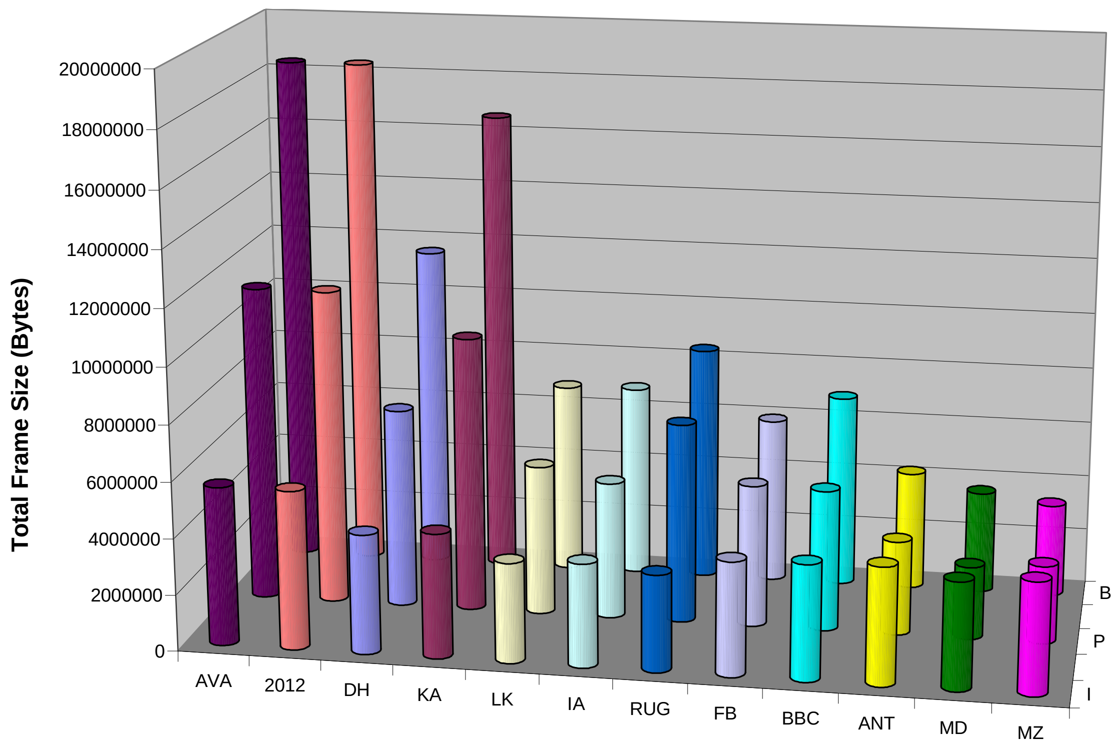
% Frame Count of All the 12 Test Clips



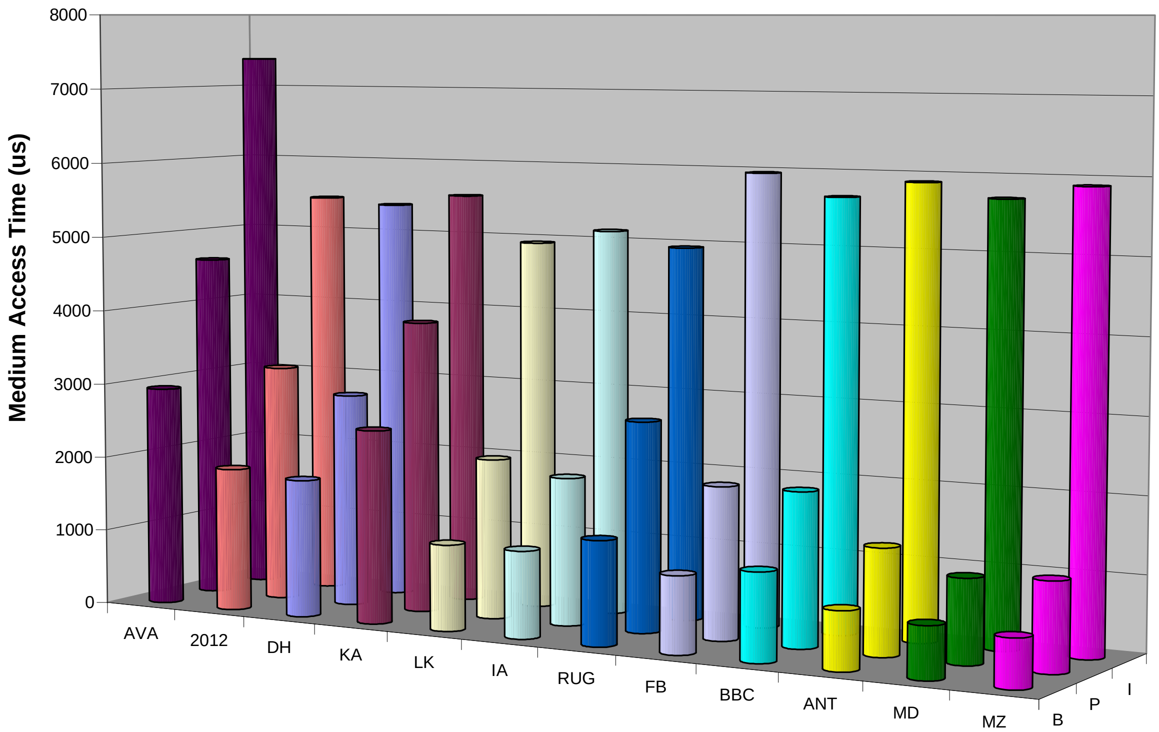
Average Frame Sizes of All the 12 Test Clips



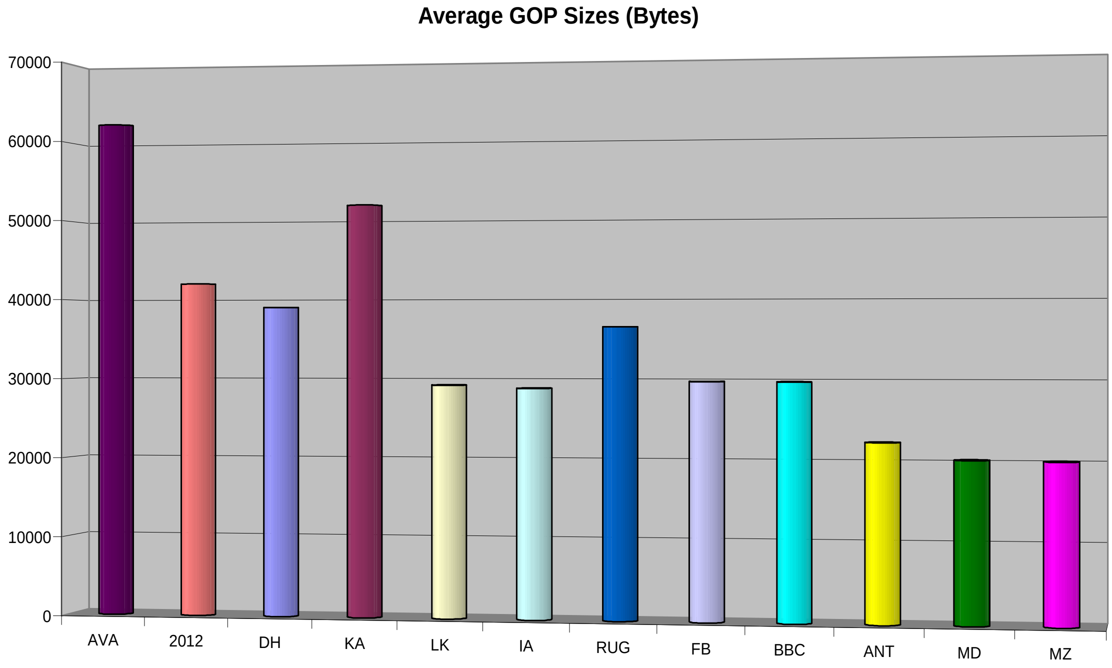
Average Frame Sizes of All the 12 Test Clips



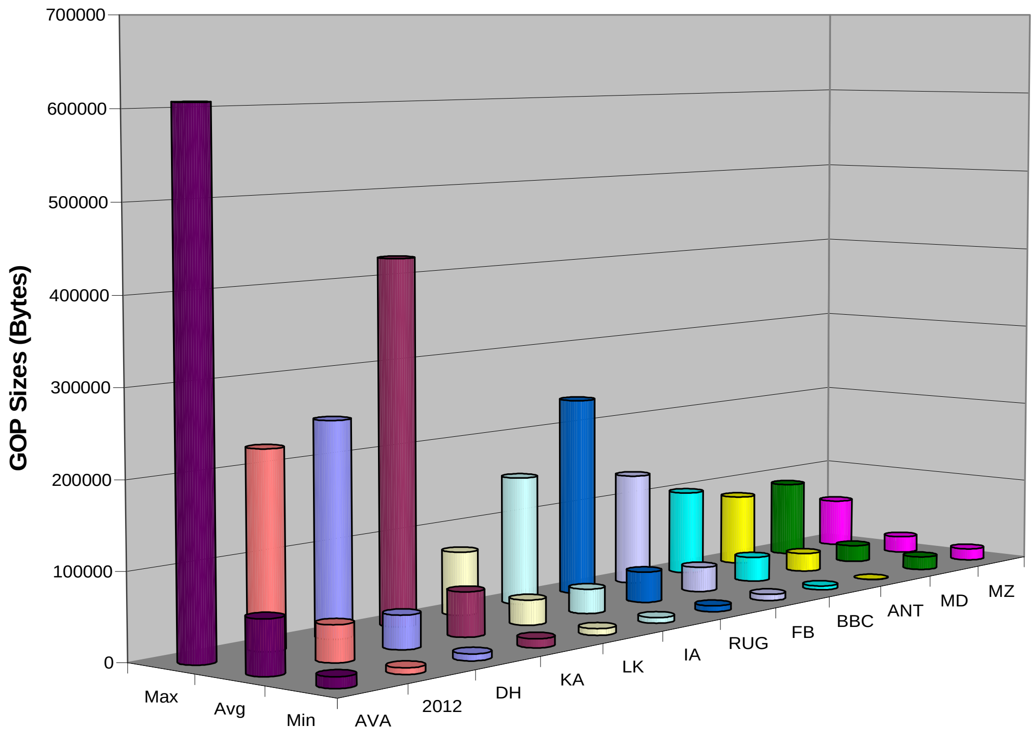
Total Frame Sizes of All the 12 Test Clips



Medium Access Time for Different Frames of All the 12 Test Clips



GOP Sizes of All the 12 Test Clips



Minimum, Maximum and Average GOP Sizes of All the 12 Test Clips

APPENDIX B RESULTS AND ANALYSIS

B.1 Scope of This Chapter

A novel QoS delivery algorithm has been proposed in chapter 4 and implemented and verified in chapter 5. It has been described in the introduction of chapter 4 that before proposing and implementing the QoS delivery algorithm different experimental scenarios were investigated and analysed to have a solid understanding of the performance of streaming video over WLAN. There are four scenarios described here. The first three are concerned with the best effort IEEE 802.11b and last one is concerned with the IEEE 802.11e WLANs. The effect of contention between stations, the effect of background traffic on streaming video, and server performance have been analyzed for an IEEE 802.11b network. The first three experimental results point out the many shortcomings of the IEEE 802.11b networks for video streaming.

Although performance can be somewhat improved as claimed by the IEEE 802.11e standard, our experimental results show that the QoS enhancement does not achieve a satisfactory level, i.e. it cannot guarantee that a single mechanism will work with all types of traffic. Thus this chapter concludes that by separating the video stream into its constituent *IPB* frames and transmitting them separately through the different ACs (i.e. queues) to obtain better performance for streamed video. This is a unique approach and is backed by experimental analysis to guarantee performance improvement to deliver video over WLANs. Based on the knowledge obtained about the relationship of *IPB* frames and video QoS, a novel QoS Delivery Algorithm is presented and validated in Chapter 4 and 5 for streamed MPEG-4 video over IEEE 802.11b WLANs.

B.1.1 The Experimental Scenarios

Scenario 1 (described Section B.3): Tests were conducted to compare the performance of a wired and wireless video server. Here frame size, frame rate, and packetisation scheme of video were varied and its effects on received bit rate, loss rate, and end-to-end packet delay were investigated.

Scenario 2 (described Section B.4): Background traffic was introduced in the second scenario. Background traffic is undesirable but unavoidable as in reality it is present in all networks in some shape or form. The effect of background traffic load on streamed video was evaluated. Tests were carried out for both the downlink and uplink load. The effect on the network in terms of loss, delay and bit rate has been studied here.

Scenario 3 (described Section B.5): Typically a WLAN network will have many clients. At a given time a number of clients will try to gain access to the shared wireless medium. Hence contention for access arises. Although IEEE 802.11b gives all competing stations equal probability of gaining access to the medium, different stations will experience different bandwidth as the capacity of the WLAN is not fixed. This important issue of contention was addressed in the third scenario.

Scenario 4 (described Section B.6): The previous scenarios have dealt with best-effort IEEE 802.11b networks where there have been no prioritizations for real time multimedia traffic. In the IEEE 802.11b real time traffic is treated in the same way as data traffic. But the IEEE 802.11e standard defines a mechanism where video and voice traffic can be given a higher priority in accessing the medium by using four tuneable parameters namely ECW_{min}, ECW_{max}, *AIFSN*, and *TXOP Limit*. By carefully tuning these parameters it is possible to enhance the network performance. The performance of parallel multimedia streaming applications under heavily loaded conditions using the *TXOP Limit* parameter was investigated. Various important factors such as delay, loss rate, throughput etc. was considered. The performance of both the audio and video streams that comprise the multimedia session was analysed.

B.1.2 Some Definitions Related to Video

Before describing the experimental results in details it is necessary to discuss some important terms for video namely frame size, hint track, inter-packet delay and frame transmission delay.

Frame Size: The video frame size is the number of packets required to transmit a single video frame and relates to the bitrate of the video frame.

Hint Track: When streaming MPEG-4 files, each video and audio track must have its own associated hint track. Hint tracks are used to support streaming by a server and indicate how the server should optimally packetize the data. The hint track MTU setting means that the packet size will not exceed in the MTU size. Thus, a hint track MTU of 512 B or 1024 B ensures that no packet for this stream will exceed 512 B or 1024 B respectively.

IPD, FTD, QFTD, PFR:

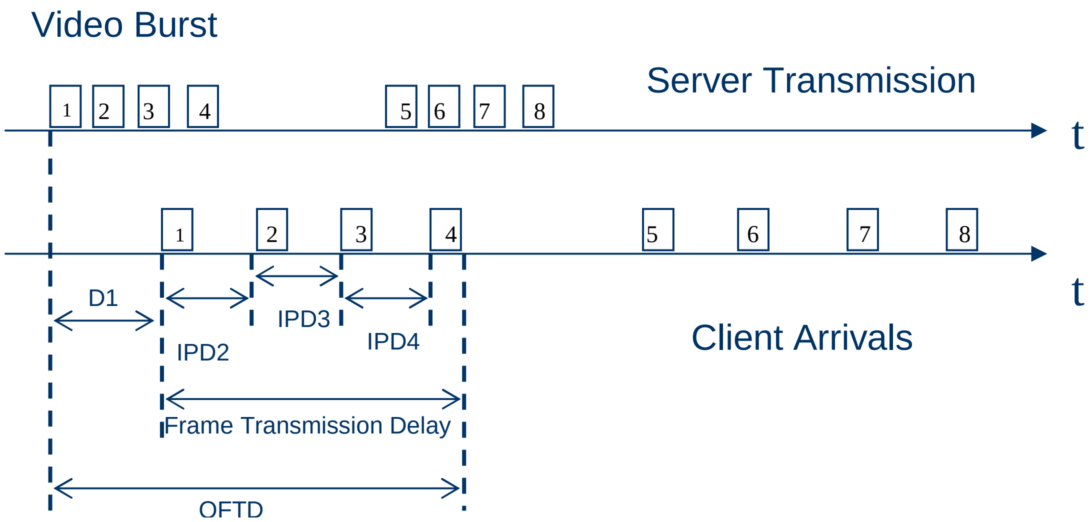


Fig. B.1: Definitions of IPD, FTD, QFTD

Inter-Packet Delay (IPD) can be defined as the difference in the measured delay between consecutive packets within a burst for a video frame at the receiver.

Frame Transmission Delay (FTD) is the end-to-end delay incurred in transmitting the entire video frame. The video frame delay is related to the number of packets required to transmit the entire video frame. The FTD is measured as the sum of the IPD for each packet required to transmit the entire video frame where the frame consists of *N* packets. The QFTD is the FTD plus the transmission delay (D) for the first packet of the video frame to reach the client.

$$FTD = \sum_{i=2}^{N} IPD_i \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad \text{(B.1)}$$

$$QFTD = D_1 + FTD \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad \text{(B.2)}$$

They are shown in Fig. B.1. In our analysis, we also consider the loss rate and the Playable Frame Rate (PFR). The PFR is inferred by using the statistical techniques described in [1].

B.1.3 On Delay and Loss of Streamed Video

Video is a frame based media and video streaming is often described as "bursty" which can have a large impact on the QoS of the video streaming application over WLAN networks. Frames are transmitted from the server to the client at regular intervals that is related to the frame rate of the video. Video with a frame rate of 25 fps will result in a frame being transmitted every 40 ms. In general, video frames are large, often exceeding the MTU of the network, and results in several packets being transmitted in a burst for each video frame. The frequency of these bursts corresponds to the frame rate of the video [2].

Delay is important for video streaming applications since all or most packets need to arrive at the client on time. Not only is the end-to-end packet delay important, but also the delay incurred when transmitting the entire video frame from the sender to the client. A video frame cannot be decoded or played out at the client until all or most of the constituent video packets for the frame are received correctly and timely. Even though WLAN networks allow for packet retransmissions, the retransmitted packet must arrive before its playout time. If the packet arrives too late for its playout time, the packet is useless and effectively lost. In a WLAN network, in addition to the propagation delay over the air, there are additional sources of delay such as queuing delays in the AP, the time required by the AP to gain access to the medium and retransmissions on the radio link layer.

When the video stream is being transmitted from the wired network to a wireless client, the arrival rate of the burst of packets is high and typically these packets are queued consecutively in the AP's transmission buffer. For each packet in the queue, the AP must gain access to the medium by deferring to a busy medium and decrementing its MAC back-off counter between packet transmissions. Since each packet must wait for the packets in the queue ahead of it to be transmitted, the end-to-end delay steadily increases until all packets in the burst have been transmitted causing the delay to vary with a sawtooth-like characteristic [3].

The duration and height of the sawtooth delay characteristic depend on the number of packets in the burst and the packet size. When there are more packets in the burst, it takes the AP longer to transmit all packets relating to this video frame.

Like delay, losses also have serious impact on the performance of video streaming applications. Loss can occur due to packets reaching their retransmission limit following repeated unsuccessful attempts and packets that are dropped due to incurring excessive delays resulting in them arriving too late to be decoded. For streamed multimedia applications, loss of packets can potentially make the presentation displeasing to the users, or in some cases make continuous playout impossible. Multimedia applications typically impose some packet loss requirements. Specifically, the packet loss ratio is required to be kept below a threshold to achieve acceptable visual quality. Packet loss ratio could be high during network congestion causing severe degradation of multimedia quality.

B.2 Experimental Tools

Windows 2000 PCs were used as server and clients. The tests reported here were all performed under Windows XP OS. Netgear Wireless cards [4] were installed in the PCs to enable them to work as wireless stations. The AP used was the Cisco Aironet 1200 model which has the IEEE 802.11b as the default setting. For IEEE 802.11e tests the firmware version *IOS 12.3(8) JA* was used which allowed users to access the IEEE 802.11e/WME capability of the device [5]. Channels were checked before each experiment to ensure that there were no other transmissions taking place.

Both simulated and real video streams were used in our tests. In the first two scenarios (described in sections B.3 and B.4) simulated video content was used. RTPTools [6] was used to mimic the sending behaviour of the video streams by enforcing the desired frame rates and burst sizes. Given the large number of encoding parameters that can be varied whilst preparing the video content for streaming over the network, only the packetisation scheme, frame rate of the video, and the size of the video frame is varied. The video frame sizes were varied between 3.1 kB, 6.1 kB and 9.2 kB. Fig. B.2 shows how the frame rate was increased every 300 sec and video frame sizes were varied every 100 sec resulting in a bitrate that increases in an Additive Increase Proportional Decrease (AIPD) manner over time and reaches a maximum bitrate of 2.1 Mbps after 1700 seconds.

Several different hint track MTU sizes were investigated. The video frame sizes were chosen to reflect the mean number of packets per video frame when using a hint track MTU setting of 1024 B and 512 B. For example, when using a hint track MTU setting of 512 B, the video frame sizes were in the {6, 12, 18} packets per video frame and when using a hint track setting of 1024 B, the video frame sizes were in the set {3, 6, 9} packets per video frame.

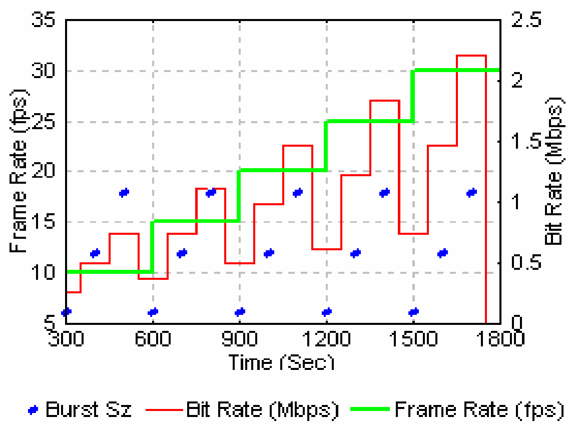


Fig. B.2: Video Stream Characteristics

In scenarios three and four (described in sections B.5 and B.6) real video content was encoded using the commercially available X4Live MPEG-4 encoder from Dicas [7]. Each content is approximately 10 minutes in duration and was encoded as MPEG-4 SP with a frame rate of 25 fps, a refresh rate of one *I* frame every 10 frames, CIF resolution and a target CBR bit-rate of 1 Mbps using 2-pass encoding. Although a target bit rate is specified, it is not always possible for an encoder to achieve this rate. Five different video content clips were used during the experiments. DH is an extract from the film ‘Die Hard’, DS is an extract from the film ‘Don’t Say a Word’, EL is an extract from the animation film ‘The Road to Eldorado’, FM is an extract from the film ‘Family Man’, and finally JR is an extract from the film ‘Jurassic Park’. The video clips were prepared for streaming by creating an associated hint track using MP4Creator from MPEG4IP [8].

In our experiments the server and client(s) were configured with WinDump [9] which is a command line tool. WinDump is the porting to the Windows platform of tcpdump [10], the most widely used network sniffer/analyzer for UNIX. WinDump can be used to monitor, diagnose and save to disk network traffic and IEEE 802.11b/g/e wireless capture. It can run under all current Windows versions - NT, 2000, XP, 2003 and Vista. WinDump captures using the WinPcap [11] library and drivers. One advantage of using Windump is that one can examine all of the traffic that moves over the network and can record any information deemed worthy of further analysis. Some commercial systems like NIKSUN's NetVCR [12] and Sandstorm’s NetIntercept [13] are also available for this purpose.

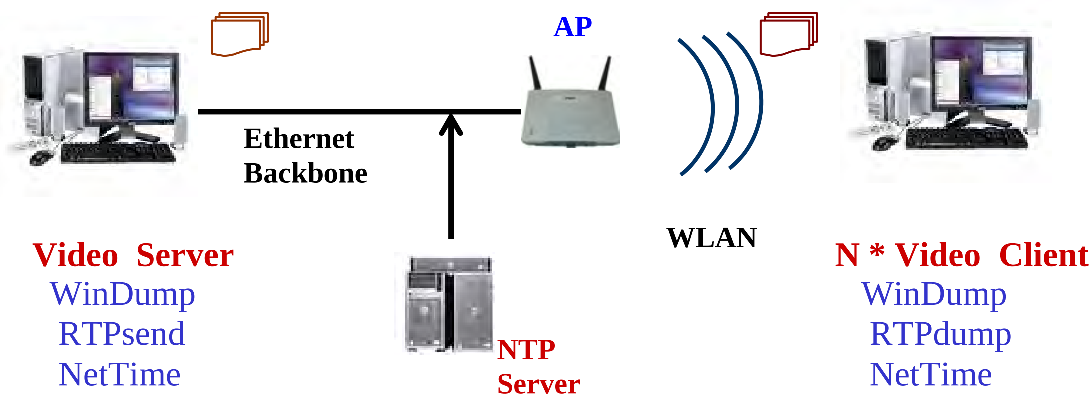
The clocks of both the client and server were synchronised before each test using NetTime [14] which is a simple time synchronization utility. Although the clocks were synchronised, a noticeable clock skew was observed in the delay measurements which was later removed using Paxson's algorithm [15,16]. The delay was measured as the difference between the time at which the packet was received at link-layer of the client and the time it was transmitted at the link-layer of the sender. MGEN [17] and DITG [18] were used to generate background traffic in different scenarios. These tools were used as they are reliable, free, and widely used by other researchers in this field. They also suited our purpose, i.e. they can be easily configured to suit our particular needs.

In one case Darwin Streaming Server (DSS) [19] and VideoLAN Client (VLC) [20] were used as a streaming server and video client respectively. There are two open source streaming servers available – one is Helix from Real [21], another is Darwin Streaming Server from Apple. DSS is an open-source, standard-based streaming server that is compliant to MPEG-4 standard profiles, ISMA streaming standards and all IETF protocols. The DSS streaming server system is a client-server architecture where both client and server consist of the RTP/UDP/IP stack with RTCP/UDP/IP to relay feedback messages between the client and server. The client can be any player that is capable of playing out MPEG-4 content. As VLC suited this profile, it was used. VLC also has the capability of recording a video content for later analysis.

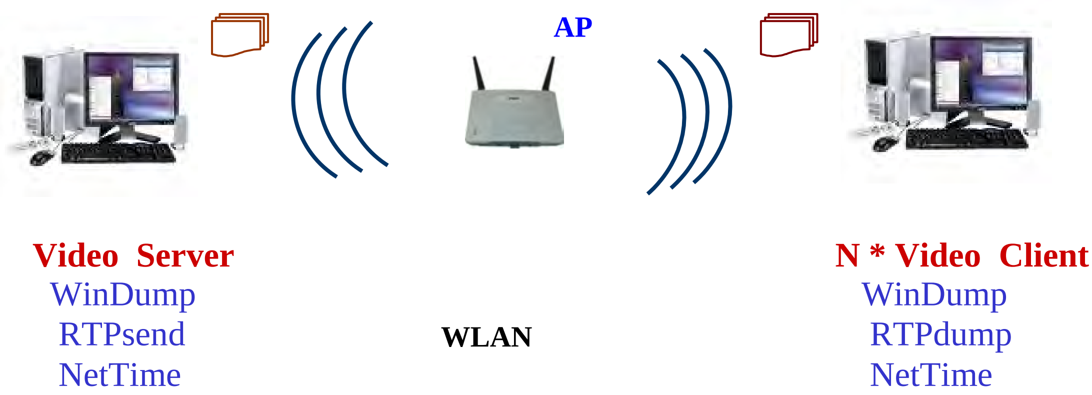
B.3 Comparison of Wired versus Wireless Video Streaming over IEEE 802.11b WLANs

Here the performance of wired and wireless servers in terms of the received bit-rate, mean packet delay, and loss rates for wired and wireless located video streaming server is compared. These results were published in the Irish Signal and Systems Conference (ISSC ’06) conference [22].

B.3.1 Experimental Testbed



(a) Wired Server



(b) Wireless Server

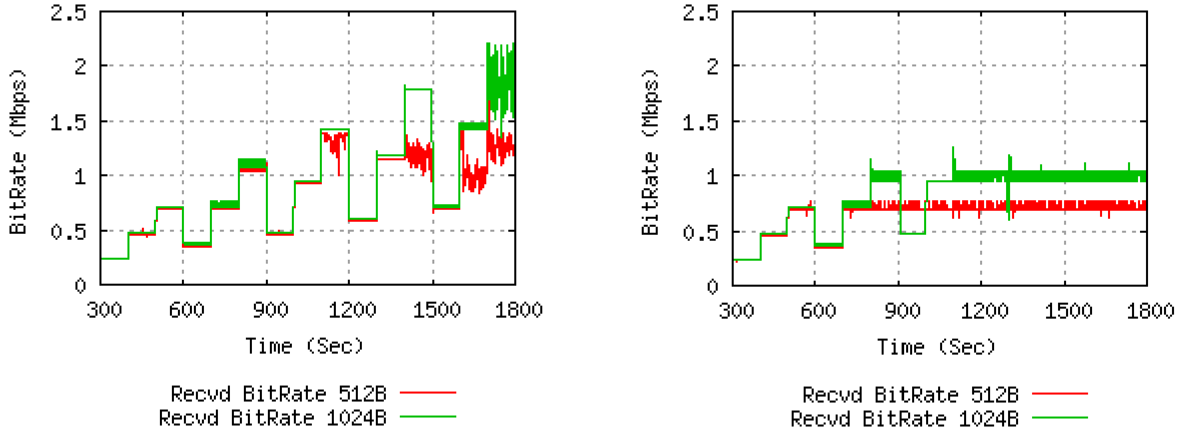
Fig. B.3: Experimental Testbed for Server Performance Evaluation

The video was streamed across the network using RTPTools. The detailed properties of the simulated video were discussed in section B.2. Both the client and server were configured with WinDump and NetTime. The clock skew observed in the delay measurements was removed using Paxson's algorithm. Two video streaming configurations for streaming MPEG-4 video are investigated as shown in Fig. B.3(a) and B.3(b). The first is when the video server is located on the wired network and is streaming video via the AP to a wireless client. The second case is when the video server is located on the WLAN and is streaming video via the AP to a wireless client.

Given that the video packets will have to gain access twice, a much poorer performance is to be expected in the second scenario. At the same time the contention generated on the network increases, so there is an increased delay on the network.

B.3.2 Capacity Analysis

To achieve an acceptable presentation quality, the transmission of a real-time video stream typically has a minimum bandwidth requirement. In this section, the received bit rate at the client is analysed. Table B.1 summarises the results for the maximum received bit rate for a wired and wireless located video server and the number of concurrent video streams using a packetisation scheme of 512 B and 1024 B. It was found that when there is a single video stream, the client receives the maximum bit rate of 2.1 Mbps



(a) **(b)**

Fig. B.4: Received Bitrate Per Client with Three Concurrent Streams for (a) Wired Located Video Server (b) Wireless Located Video Server

Table B4.1: Comparison of Received Bit-Rates

	1 Video Client Maximum Received Bit Rate (Mbps)				2 Video Clients Maximum Received Bit Rate (Mbps)				3 Video Clients Maximum Received Bit Rate (Mbps)			
	512B		1024B		512B		1024B		512B		1024B	
	Per Client	Total Recvd Load	Per Client	Total Recvd Load	Per Client	Total Recvd Load	Per Client	Total Recvd Load	Per Client	Total Recvd Load	Per Client	Total Recvd Load
Wired Server	2.10	2.10	2.10	2.10	2.05	4.10	2.10	4.20	1.3	3.90	2.00	6.00
Wireless Server	2.10	2.10	2.10	2.10	1.10	2.20	1.50	3.00	0.75	2.25	1.00	3.00

from the video server located in the wired network regardless of the packetisation scheme used. However as the number of concurrent video streams is increased, the packetisation scheme reduces the received bit rate. When the number of concurrent video streams is increased to two and three streams, the received bit rate by each client is reduced to 2.05 Mbps and 1.3 Mbps respectively when using a packetisation scheme of 512 B. However, when using a packetisation scheme of 1024 B, each client receives the maximum bit rate of 2.1 Mbps and 2.0 Mbps respectively. A similar trend is observed when using a wirelessly located video streaming server. When the server is using a packetisation scheme of 512 B, the maximum received bit rate per client is reduced from 2.1 Mbps to 1.1 Mbps to just 0.75 Mbps as the number of concurrent video streams is increased from one to three. Similarly, when using an MTU of 1024 B, the maximum received bit rate per station is reduced from 2.1 Mbps, to 1.5 Mbps to 1 Mbps.

Fig. B.4 shows the received bit rate for a wired and wireless server with 3 concurrent streams. It can be seen that the WLAN becomes saturated when there are three concurrent streams. When using a wired server, the AP becomes saturated with a total throughput of 6 Mbps and 3.9 Mbps when using a packetisation scheme of 1024 B and 512 B respectively. The wireless located server achieves a maximum throughput of 3 Mbps using 1024B packetisation scheme and 2.25 Mbps using 512 B packetisation scheme. The maximum received bit rate is less when using a smaller packetisation scheme. When using a smaller packet size, more packets are required to transmit the same amount of video data. The AP must gain access to the medium to transmit each packet by deferring to a busy medium and decrementing its MAC back-off counter between packet transmissions. For 512 B packets the AP must gain access to the medium twice as often compared to 1024 B packets which increases the likelihood of collisions and packets being dropped at the AP queue so the received bit rate was less when using 512 B packets. However by using larger packets, the AP accesses the medium and transmits the data more efficiently, i.e. it makes more efficient use of its transmission opportunities.

The received bit rate was always less when using a wireless located server than that achieved for wired server for multiple clients. When both the server and client are located on the same WLAN, the video stream occupies twice as much resources since the video is transmitted from the server to the AP and then from the AP to the video client. For example, it can be seen that when there are three concurrent streams using 1024 B packetisation, the WLAN becomes saturated at 6 Mbps using a wired server and 3 Mbps using a wireless server. However given that the wireless server uses twice as many resources to transmit on the uplink to the AP and on the downlink to the client, the stream in fact occupies 6 Mbps.

B.3.3 Loss Rate Analysis

Here the bit rate of the video stream increases over time. As a consequence the loss rate of the video stream varies over time. Fig. B.5(a) and Fig. B.5(b) show the loss rate variations for a wired video server for one to three concurrent video streams using a packetisation scheme of 512 B and 1024 B. It can be seen that when there are three concurrent video streams, the loss rates reach 30% and 15% when the bit rate reaches a maximum for a packetisation scheme of 512 B and 1024 B. By using a packetisation scheme of 512 B, twice as many packets are required to transmit the video frame. In this way, the transmission buffer at the AP becomes saturated more quickly resulting in packets being dropped.

In contrast when using a wireless video server, as shown in Fig. B.5(c) and Fig. B.5(d), the loss rates remain at relatively low levels at less than 1% but are throughout the experiments. Loss in the WLAN medium occurs due to collisions and packet retransmissions. Packets are lost when they reach their retransmission limit. It can be seen that when using a smaller packet size, there is a higher loss rate and this is due to the increased number of packets that need to be transmitted. It can also been seen that the number of concurrent streams does not affect the observed loss rates.

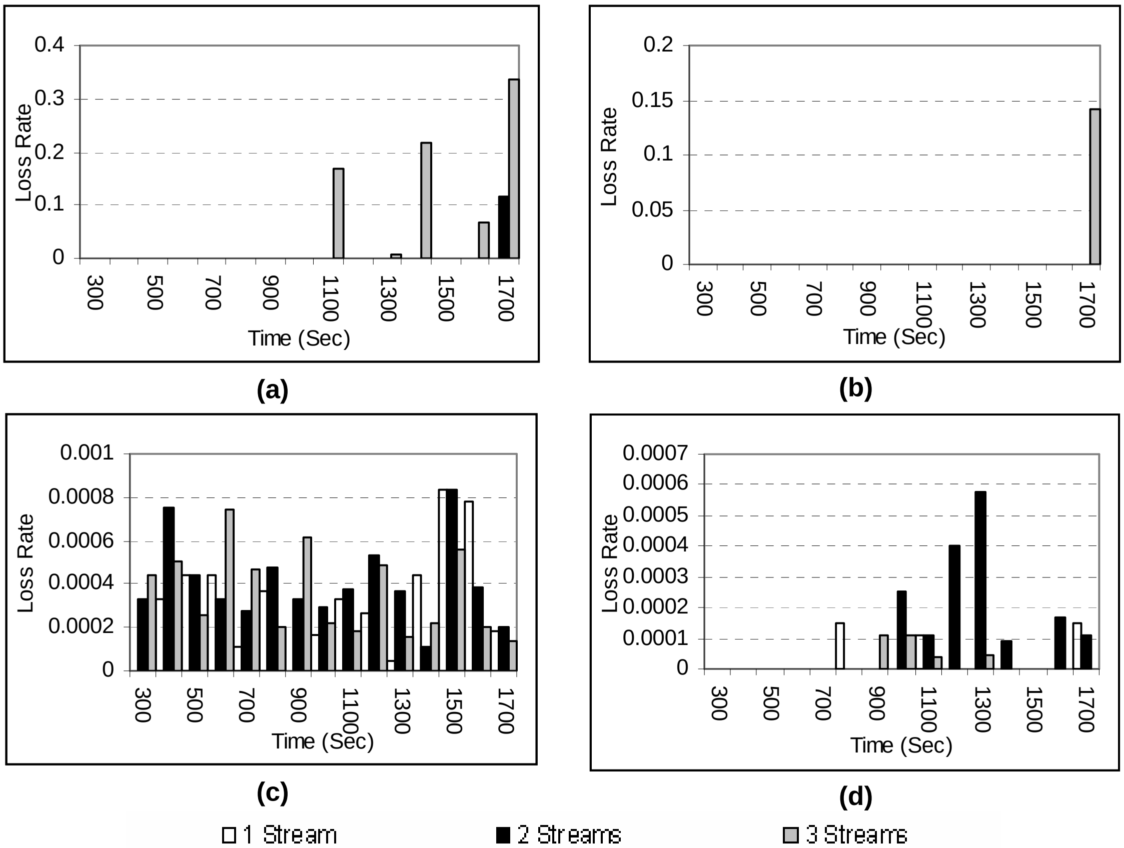


Fig. B.5: Loss-Rate for 3 Concurrent Video Streams
(a) Wired Server Using 512 B Packetisation Scheme,
(b) Wired Server Using 1024 B Packetisation Scheme,
(c) Wireless Server Using 512 B Packetisation Scheme,
(d) Wireless Server Using 1024 B Packetisation Scheme

B.3.4 Delay Analysis

Fig. B.6(a-d) shows how the mean network delay averaged every second varies over time for streaming the video clip MTU setting of 1024 B and 512 B respectively for one to three concurrent video streams. In the experiments reported here, the size of the video frame is increased every 100 sec. Fig. B.6(a) shows the delay variations over time for a wireless video server using a packetisation scheme of 1024 B for one to three concurrent video streams. It can be seen that as the number of video streams is increased, the mean delay is increased since there are more packets in the AP transmission buffer and so the packet must wait longer in order to be transmitted.

In addition, the mean delay is affected by the packetisation scheme used as can be seen by comparing Fig. B.6(a) and Fig. B.6(b). This is expected since the smaller the packet size, the greater the number of packets that are in the queue at the AP. With a greater number of packets in the queue, the video packets are more likely to be delayed longer since they must wait for the AP to gain access to the medium by deferring to a busy medium and decrementing its back-off counter for each of the packets in the queue ahead of it.

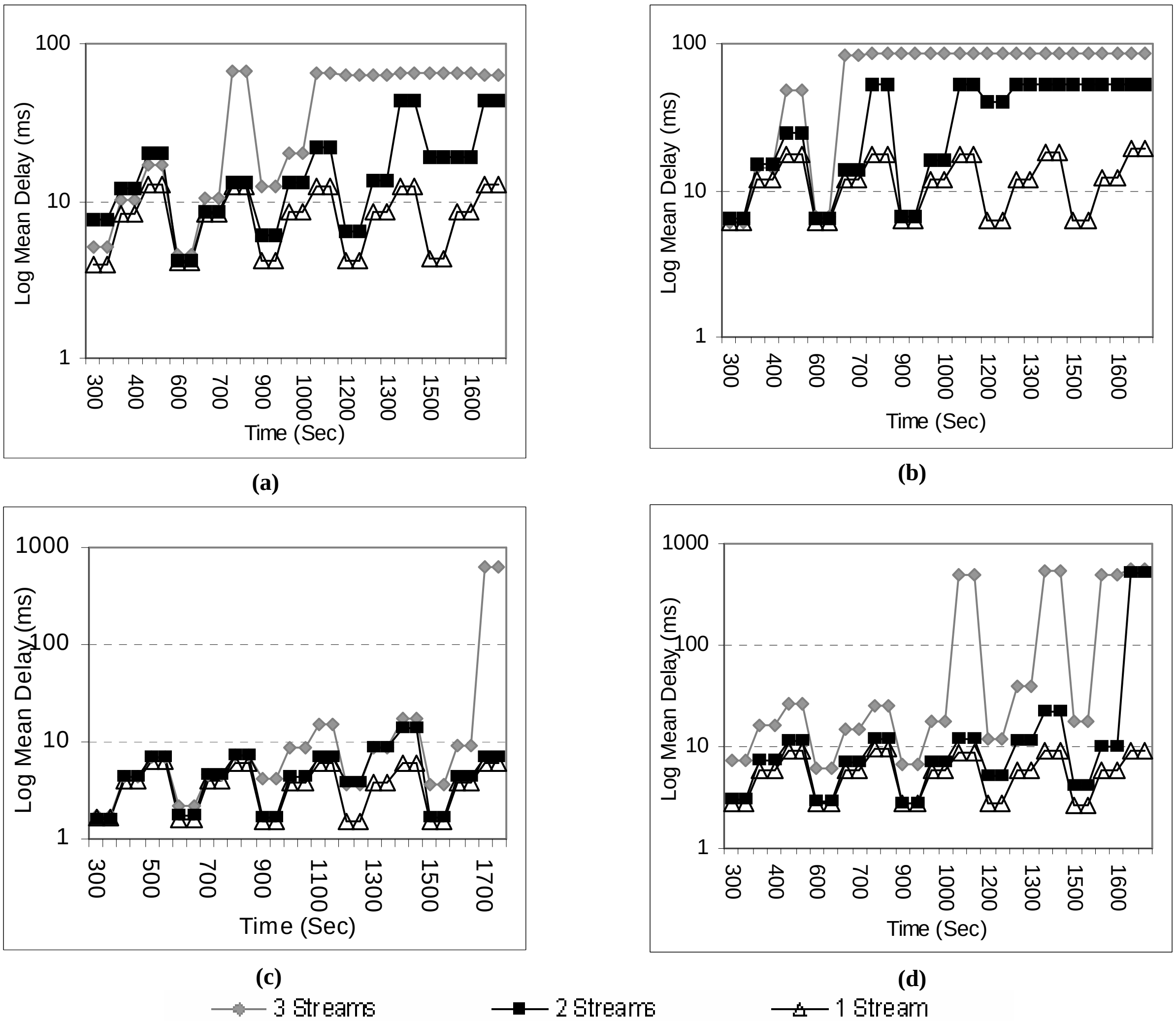


Fig. B.6: Mean Delay for Wired And Wireless Located Video Server (a) Wireless Server Using 1024 B Packetisation Scheme (b) Wireless Server Using 512 B Packetisation Scheme (c) Wired Server Using 1024 B Packetisation Scheme (d) Wired Server Using 512 B Packetisation Scheme

The mean delay is closely related to the size of the video frame. For example, if many packets are required to send the video frame, the AP must access the medium in order to transmit each packet and so each packet must wait longer in the AP transmission buffer causing it to experience increased delays. This can be seen by comparing the delay variations for three concurrent streams in Fig. B.6 (a) and Fig. B.6 (b) with Fig. B.6 (b) that shows the maximum received bit rate.

Despite a few spurious results for the case of 3 streams (i.e. in Fig B.6 (c) and Fig. B.6 (d)), the mean delay experienced by the wired servers is less than that of wireless servers. It is believed that these spurious results are due to external interference from other WLAN users within the building where the experiments were performed.

B.3.5 Conclusions

We compared the performance of wired and wireless video streaming for two different packetisation schemes in terms of bit rate, loss rate and packet delay.

It was found that the received bit rate was much higher when using a wired server and large packetisation scheme. However, this can be traded off against an increased packet loss rate when there are many concurrent streams. The wireless server has a higher packet delay and lower loss rates. Also the packetisation scheme has an important effect on all these parameters. By using small packets not only is there an increased header overhead due to the fact that more packets are required to send the same amount of data, but also more MAC layer ACKs need to be sent. In addition, by using small packets the AP must access the medium more often which results in packets incurring greater queuing delays. In addition, due to the increased queuing delays, it is more likely that the AP transmission buffer will become saturated which can result in packets being dropped under heavily loaded conditions.

B.4 Effects of Background Traffic Loads on Streamed Video over IEEE 802.11b WLANs

In the previous section the performance of wired and wireless servers in the absence of any background traffic was compared. Next, the impact of background load on the performance of streaming MPEG-4 video is investigated. The performance for both uplink and downlink background loads is analyzed. The performance is measured in terms of the key parameters of bit rate, loss rate and mean delay since these are the primary factors that affect the perceived video quality at the receiver. The results described in this section were published in the Information Technology and Telecommunications (IT&T) conference [23].

B.4.1 Experimental Testbed

In the experiments a video streaming session was established between the video client and server and the background traffic load was sent from uplink and separately then from downlink (Fig. B.7). When the background traffic load originates on the wired network and is transmitted to a wireless background sink station via the AP, it is referred to as a *Down-Link Load (DLL)*. The second case is when the background load originates in the WLAN where each background load station generates a load of 1 Mbps and is transmitted towards a wired sink station, this will be referred to as an *Up-Link Load (ULL).*

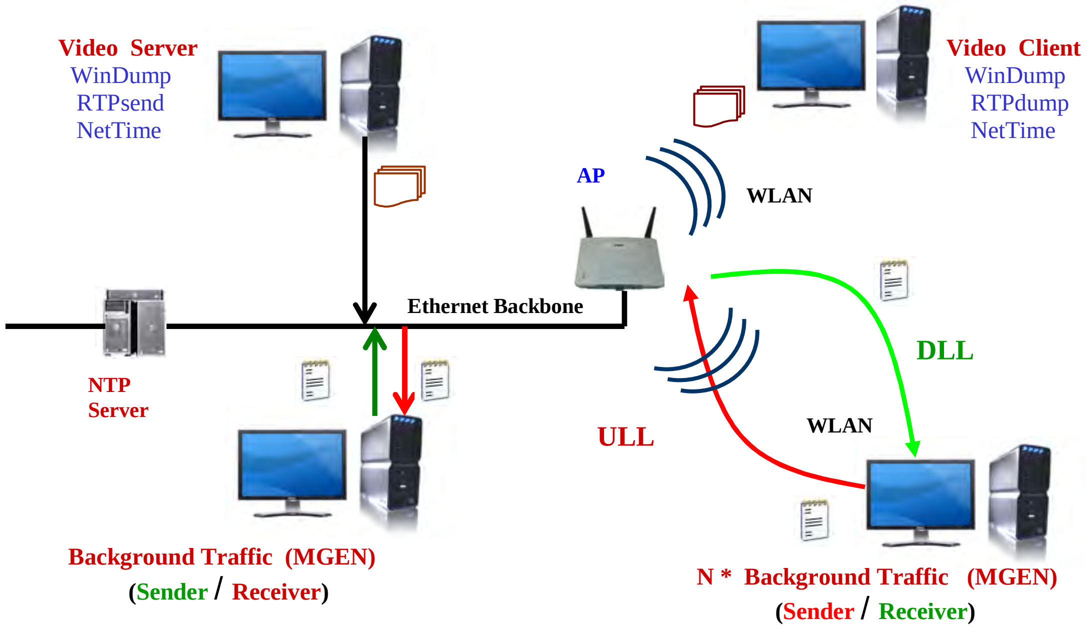


Fig. B.7: Experimental Testbed for Background Traffic Scenario

For example, to generate an ULL of 5 Mbps there were 5 wireless background stations each with an offered load of 1 Mbps. The background traffic was generated using MGEN. In our experiments, the DLL and ULL load was kept constant throughout the duration of the video streaming session and transmitted using 1024 B packets.

The video was streamed across the network using RTPTools. The detailed properties of the simulated video were discussed in the section B.2. Windump, NetTime and Paxon's algorithm were used for packet monitoring, synchronization and skew removal in delay measurements respectively.

B.4.2 Results

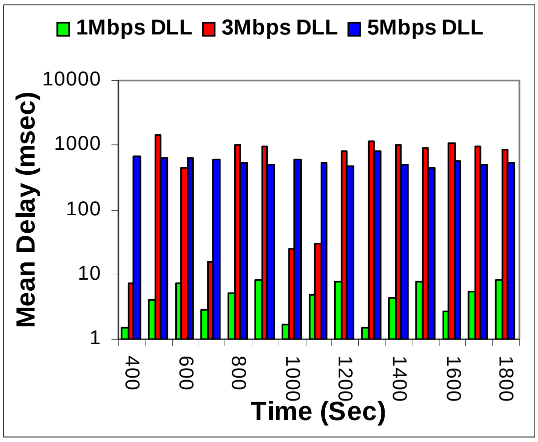
Here we compare the performance of the video streaming session in terms of the mean packet delay, received bit rates, and loss rates with increasing levels of DLL and ULL. Video stream is affected differently by an ULL and DLL. In addition, we show that the performance of the video streaming application is also dependent on the bit rate of the video and the number of packets required transmitting each video frame.

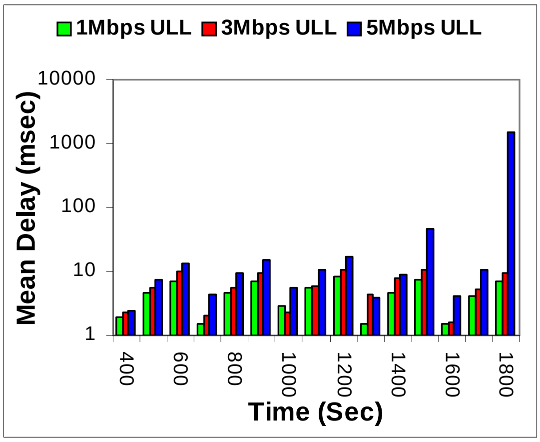
B.4.2.1 Delay

The packet delay for the video stream packets include the queuing delay for the packet to reach the head of the AP transmission buffer, the time required for the AP to win a transmission opportunity, *DIFS*, *SIFS*, propagation delays and the time for the MAC Acknowledgement to be received.

In Fig. B.8 (a) it can be clearly seen that when there is a DLL of 1 Mbps, the mean packet delay is less than 10 ms. However, with a DLL of 3 Mbps and 5 Mbps the mean packet delay exceeds 500 ms. With this increased DLL the AP must serve not only the video stream packets but also the background traffic. With increased buffer occupancy, there are increased queuing delays for the video stream.

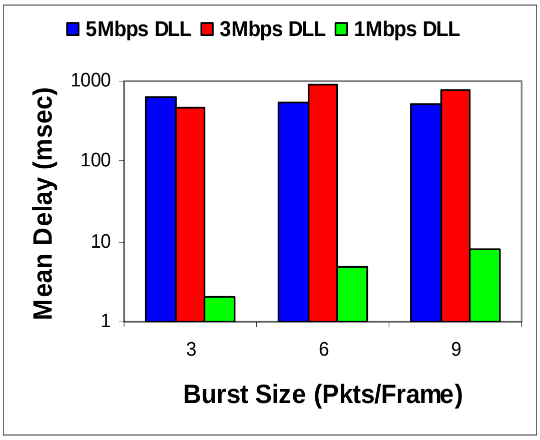
In contrast, Fig. B.8 (b) shows the mean packet delays with an increased ULL. It can be seen that the mean packet delay is approximately 10 ms regardless of the offered ULL. The reason for this difference is that with ULL background traffic, packets are not queued at the AP transmission buffer since they are destined for a wired sink station. This means that the video packets do not experience an increased queuing delay.

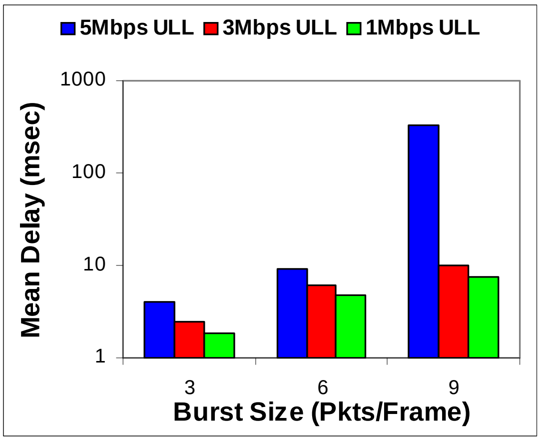




B.8 (a)

B.8 (b)





B.8 (c)

B.8 (d)

Fig. B.8: Mean Delay with Increasing Background Load (a) DLL and (b) ULL; and Burst Size (c) DLL and (d) ULL.

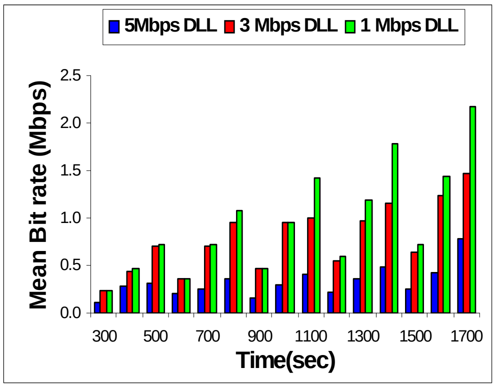
In addition, during the video streaming session the bit rate and frame rate of the video stream is steadily increasing which results in an increased number of packets required to transmit each video frame, i.e. the burst size. It can be seen in Fig. B.8 (c) and Fig. B.8 (d) that there is a clear relationship between the mean packet delay and the burst size. This is expected since if there are more packets required to transmit the video frame, the mean packet delay will be increased. Table B.2 summarises the mean packet delays for a DLL and ULL. For 3 Mbps DLL background load and a burst size of 6, a large delay (889.1 ms) was observed. It can be identified as a spurious result arising from interference from other WLAN users in the neighborhood.

Table B.2: Average Delay for Different Burst Sizes for Different Background Loads (milliseconds)

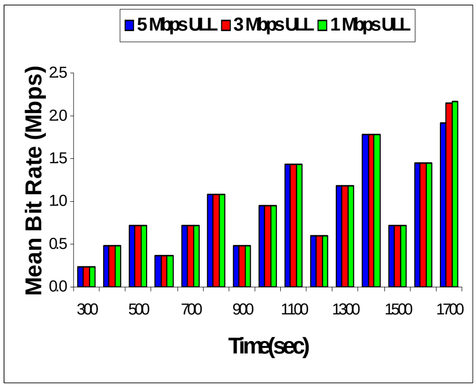
	Burst Size (pkts/frame)	Background Load		
		5 Mbps	3 Mbps	1 Mbps
DLL	3	648.00	457.23	2.03
	6	543.18	889.10	4.78
	9	524.25	787.19	7.86
ULL	3	4.03	2.52	1.86
	6	9.39	6.01	4.77
	9	325.08	9.95	7.46

B.4.2.2 Capacity Analysis

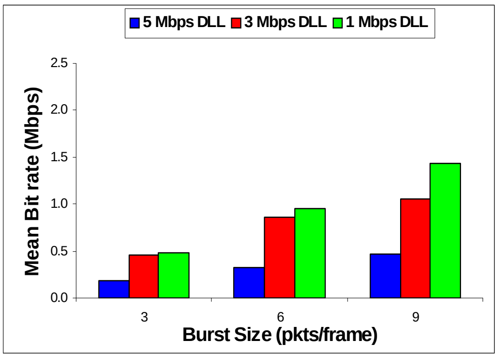
The experimental results presented in Fig. B.9(a) and Fig. B.9 (c) show that the received bit rate at the client is higher with a 1 Mbps DLL than with a 5 Mbps DLL. The increase in burst size refers to an increase of the number of packets to be transmitted. The reason for the differences in the received bit rate for a DLL of 5 Mbps is that the AP has become saturated. The AP must then transmit up to 2.2 Mbps of video stream plus 5 Mbps of background traffic. As a result of saturation, packets are being dropped from the AP buffer and the received bit rate continues to decrease as the amount of background traffic increases.



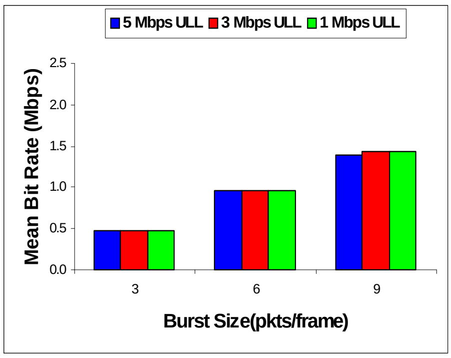
B.9 (a)



B.9 (b)



B.9 (c)



B.9 (d)

Fig. B.9: Mean Bitrate with Increasing Background Load

(a) DLL and (b) ULL; and Burst Size (c) DLL and (d) ULL.

However, in Fig. B.9 (b) and Fig. B.9 (d) it can be seen that the received bit rate is unaffected by the ULL. The ULL background traffic packets are not queued at the AP transmission buffer since they are destined for a wired sink station. This means that the video packets do not experience the effect of background load present and the only packets queued in the AP buffer belong to the video stream and so the AP does not become saturated. So irrespective of offered load the received mean bit-rate was similar for a certain burst size and ULL. Table B.3 summarises the average received bit rate for a DLL and ULL.

Table B.3: Average Received Bit Rate (Mbps) at The Client

	Burst Sz (pkts/frame)	Background Load		
		5 Mbps	3 Mbps	1 Mbps
DLL	3	0.18	0.45	0.48
	6	0.32	0.86	0.96
	9	0.47	1.05	1.44
ULL	3	0.48	0.48	0.48
	6	0.96	0.96	0.96
	9	1.39	1.43	1.43

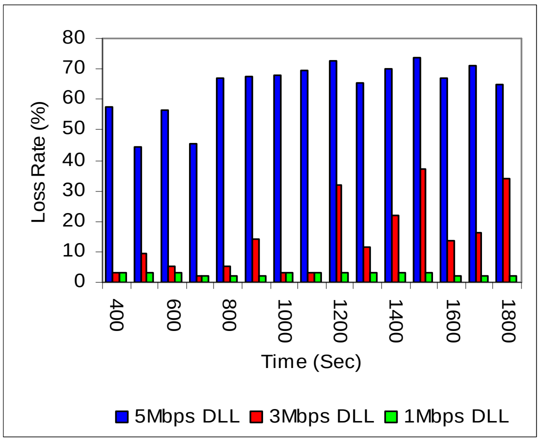
B.4.2.3 Loss Rate

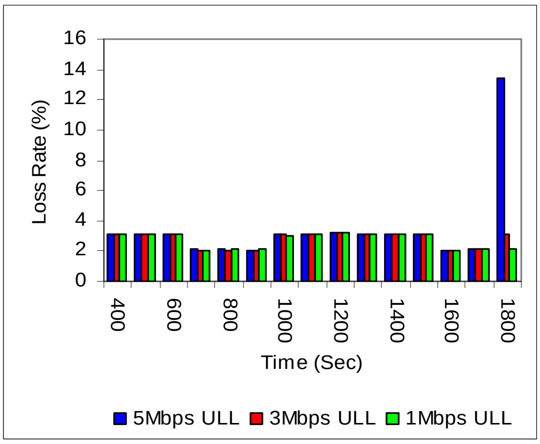
Table B.4 summarizes the average loss rate for a DLL and ULL.

Table B.4: Average Loss Rate (%) at The Client

	Burst Size (pkts/frame)	Background Load		
		5 Mbps	3 Mbps	1 Mbps
DLL	3	60.55	6.72	2.69
	6	64.41	11.24	2.73
	9	67.02	24.52	2.72
ULL	3	2.71	2.71	2.67
	6	2.73	2.71	2.73
	9	4.97	2.94	2.74

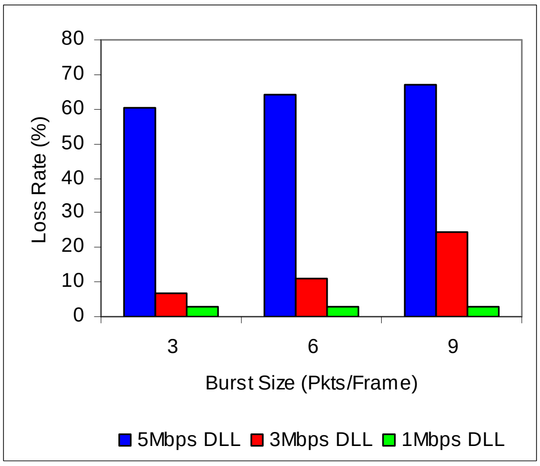
There is a relationship between the received bit rate and loss rate. Fig. B.10(a) shows the loss rates incurred by the video stream with a DLL. It can be seen that there is a significant loss rate with a DLL of 5 Mbps. This again can be explained by the increased occupancy at the AP buffer causing packets to be dropped as the AP buffer can hold only a finite number of packets. With such high loss rates, the video stream is unwatchable. In contrast Fig. B.10 (b) shows the loss rates with an increased ULL. The loss rates are of the order of 2% which will have a minimal affect on the decoding of the video stream since video applications can tolerate small loss rates. Fig. B.10 (c) and Fig. B.10 (d) show the effect of burst sizes on loss rates. As the burst size is steadily increasing over time the number of packets required to transmit the video frame is greater.

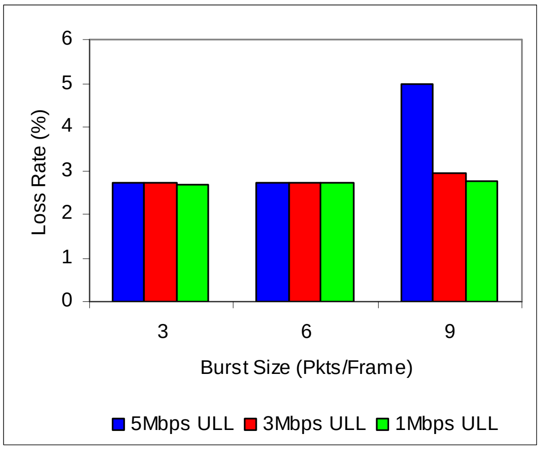




B.10 (a)

B.10 (b)





B.10 (c)

B.10 (d)

Fig. B.10: Mean Loss Rate with Increasing Background Load (a) DLL and (b) ULL; and Burst Size (c) DLL and (d) ULL

For the DLL load, the AP becomes saturated as it has to transmit both the video and background packets. This eventually leads to an increase in the mean packet delay which in turn increases the loss rate as more packets get lost due to congestion. It can be seen in Fig. B.10 (c) that for the 5 Mbps background load the loss rate was more than 60%. For ULL a small loss was observed in Fig. B.10 (d), as only the video packets were transmitted.

B.4.3 Conclusion

Experimental results show that the performance of the video stream is more severely affected with a DLL than with an ULL. The mean packet delay was much lower for an ULL as there are fewer queuing delays in the AP transmission buffer. Furthermore, it is evident that the delay is related to the number of packets required to transmit the video frame, i.e. the burst size. The received bit rate is greater for an ULL than for a DLL since with a DLL there is higher buffer occupancy at the AP which leads to packets being dropped at the AP buffer. With such packet drops, the loss rate also increases. Thus it can be seen that the performance of the video stream is significantly reduced with a DLL than with an ULL. However, it should be noted that in our experiments for an ULL, only a small number of stations were used which reduced the effects of contention on the video stream. It is expected that with a large number of ULL stations, the AP should experience an increased number of deferrals due to contention with other stations which in turn increases the time it takes the AP to win a transmission opportunity.

B.5 The Effects of Contention Between Stations on Video Streaming Applications

We now turn our attention to another important feature in a WLAN environment - contention between stations. We experimentally analyse the effects of contention on the performance of video streaming applications. Keeping the total offered load in the network constant with varying contention levels we demonstrate the effect on the frame transmission delay. The results described in this section were published in the Information Technology and Telecommunications (IT&T) conference [24].

B.5.1 Video Content Preparation and Analysis

The properties of the five types of encoded clips were discussed in section B.2. It is necessary to repeat the experiments for a number of different video content types since the characteristics of the streamed video have a direct impact on its performance in the network. Each video clip has its own unique signature of scene changes and transitions which affect the time varying bit rate of the video stream. Animated videos are particularly challenging for encoders since they generally consist of line art and as such have greater spatial complexity.

Table B.3 summarizes the characteristics of the encoded video clips used during the experiments. The second column shows the mean packet size of the clip as it is streamed over the network and the third column shows the mean bit-rate of the video clip. The following columns show the maximum video frame size and the mean video frame size in bytes as measured over all frames, over *I* frames only and *P* frames only. Finally, the last column shows the peak-to-mean ratio of the video frames. It can be seen that despite encoding the video clips with the same video encoding parameters, the video clips have different characteristics. Despite all the video clips being prepared with exactly same encoding configuration, due to the content of the video clips the mean and maximum *I* and *P* frames vary considerably in size.

Table B.3: Characteristics of Encoded Video Clips

Clip	Mean Packet Size (B)	Mean Bit Rate (kbps)	Frame Size (B)		*I* Frame Size (B)		*P* Frame Size (B)		Peak-to-Mean Ratio
			Max.	Avg.	Max.	Avg.	Max.	Avg.	
DH	889	910	16762	4617	16762	7019	12783	812	3.63
DS	861	682	12734	3480	12734	6386	10600	713	3.66
EL	909	1199	27517	6058	27517	14082	14632	1587	4.54
FM	894	965	17449	4903	17449	10633	15078	1188	3.56
JR	903	1081	17299	5481	17299	8991	13279	1006	3.16

B.5.2 Experimental Testbed

To demonstrate the effects of station contention on video streaming applications, the video server was set up on the wired network and streamed the video content to a wireless client via the AP (Fig. B.11). The video streaming system consists of the Darwin Streaming Server (DSS) acting as the video server and VideoLAN Client (VLC) as the video client. We used

Windump, NetTime and Paxon's algorithm for packet monitoring, synchronization, and skew removal respectively.

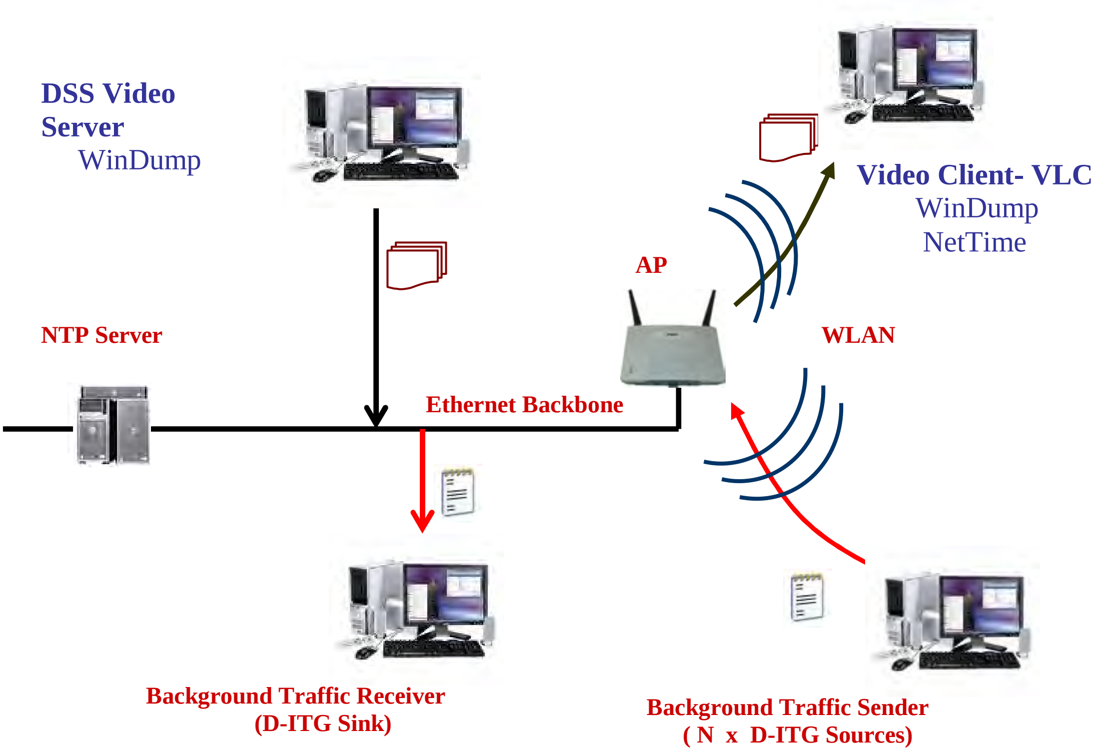


Fig. B.11: Experimental Setup to Evaluate Contention

There are a number of wireless background load stations contending for access to the WLAN medium where their traffic load is directed towards a sink station on the wired network. The background uplink traffic was generated using Distributed Internet Traffic Generator (D-ITG). The background traffic load has an exponentially distributed inter-packet time and an exponentially distributed packet size with a mean packet size of 1024B. To maintain a constant total background load of 6 Mbps, the mean rate of each background station was appropriately decreased as the number of background stations was increased.

B.5.3 Results

When there are no other stations contending for access to the medium, the IPD is in the range 0.9 ms to 1.6 ms for 1024 B sized packets. This delay range includes the *DIFS* and *SIFS* intervals, data transmission time including the MAC Acknowledgement as well as the randomly chosen BC values of the IEEE 802.11 MAC mechanisms contention windows in the range 0-31. This can be seen in Fig. B.12 which shows that probability density function (PDF) of the IPD with and without contention. It can be seen that there is an upper plateau comprising 32 peaks corresponding to each of the possible 32 BC values with a secondary lower plateau that corresponds to the proportion of packets that were required to be retransmitted through subsequent doubling of the contention window under the exponential binary backoff mechanism employed in the IEEE 802.11 MAC.

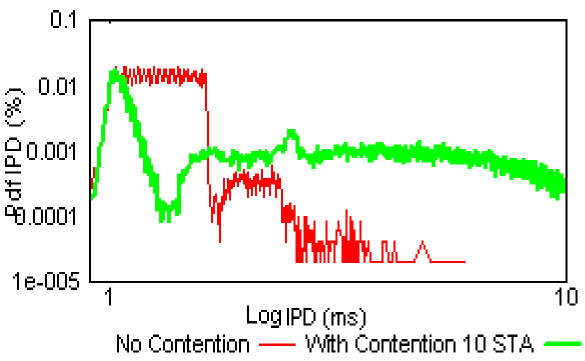


Fig. B.12: The PDF of The IPD With and Without Contention

As contention levels increase, all stations must pause decrementing their BCs' more often when another station is transmitting on the medium. As the level of contention increases, it takes longer to win a transmission opportunity and consequently the maximum achievable service rate is reduced which increases the probability of buffer overflow. In these experiments, the nature of the arrivals into the buffer remains constant, i.e. only the video stream is filling the AP's transmission buffer with packets. By varying the number of contending stations we can affect the service rate of the buffer and thereby its ability to manage the burstiness of the video stream. This can be seen in Fig. B.12 where there is a long tail in the distribution of IPD values for the 10 station case. In this case, 10 wireless background traffic stations are transmitting packets to the wired network via the AP's receiver. The aggregate load from these stations is held constant as the number of background stations is increased.

B.5.3.1 The Effects of Contention

Table B.4: Mean Performance Values for DH Clip with Increased Contention (DC = 500ms)

Performance Metric	0STA	3STA	4STA	5STA	6STA	7STA	8STA	9STA	10STA
Mean Delay	10.43	29.62	30.97	37.91	63.63	105.75	174.91	311.71	395.27
FTD	11.50	36.62	37.96	45.39	71.76	115.61	186.05	325.01	406.83
IPD	1.24	3.73	3.75	3.97	4.34	4.82	5.27	5.66	5.95
Mean Loss Rate (DC > 500ms)	0.00	0.01	0.01	0.03	0.08	0.15	0.23	0.34	0.41
PFR (fps) (DC > 500ms)	25.00	25.00	23.00	21.83	19.04	16.91	14.02	10.51	9.92

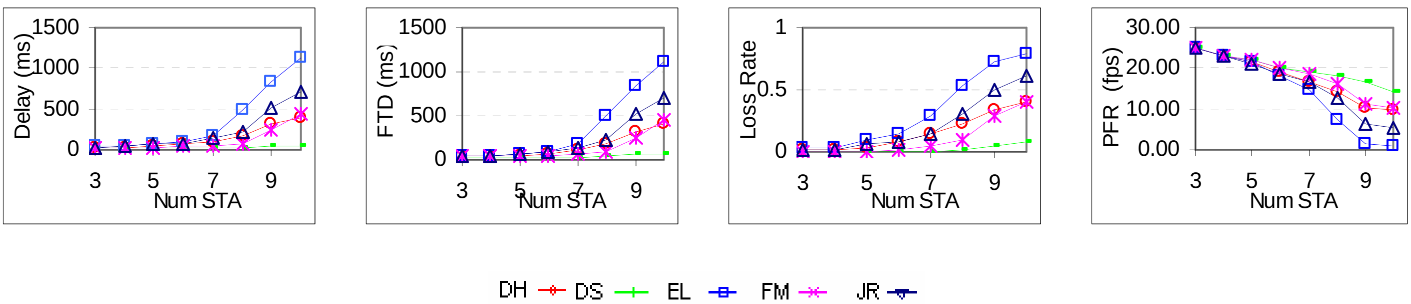


Fig. B.13: Mean Values for A Number of Video Clips for a Fixed Total Offered Uplink Load with Increased Number of Contributing Stations (a) Mean Delay, (b) Mean FTD, (c) Average Loss Rate with a DC (delay constraint) of 500ms, (d) Inferred PFR with A DC of 500ms.

In this section, the effects of contention on video streamed applications are experimentally demonstrated. A single video clip DH was streamed from the wired network via the AP to a wireless client. This particular clip was chosen since it is representative of a typical non-synthetic video stream. Table B.4 presents the mean performance values for the video clip DH over the test period with increased contention. It can be seen that the mean delay, loss rate, FTD and IPD increase with increased contention. In this work we have set the DC (Delay Constraint) to 500 ms which is the delay constraint for low latency real-time interactive video.

It can be seen that when there are no background contending stations, the mean packet delay is about 10 ms. As the number of contending stations increases from 3 to 7 to 10, the mean delay increases from 30 ms, to 100 ms, and to 400 ms respectively. As the number of contending stations is increased from 3 to 7 to 10 stations with a DC of 500 ms, the mean loss rate including packets dropped due excessive delay is increased from 1% to 15% to 41% respectively. This in turn affects the ability of the codec to decode the video frames since there is increased likelihood that packets will not arrive within the given delay constraint.

The experiment was repeated for the other video clips all encoded with the same encoding configuration but having different content complexity characteristics. For all content types, it can be seen that the mean packet delay and FTD increase with increased contention as shown in Fig. B.13(a) and Fig. B.13 (b). Fig. B.13 (c) shows the mean loss rate over the test period for each of the video clips where it can be seen that there is a dramatic increase in the mean loss rate when the number of contending stations exceeds 7 stations when a delay constraint of 500 ms is imposed on the system which results in an even greater impact of the contention on performance. Fig. B.13 (d) shows the PFR that is statistically inferred from the packet loss and delay. Apart from the impact of contention, Fig. B.13 (a) to Fig. B.13 (d) also highlight the impact of the video content where it can be seen that the animation clip *EL* is the most severely affected by increased contention whilst the clip *DS* is the least affected. The high complexity of the animation clip *EL* is due to frequent scene cuts and line art within the scene that affects the burstiness of the encoded video sequence since much more information is required to encode the increased scene complexity.

B.5.3.2 Analysis

Here we generalize the results presented in the previous section to account for all content types and a given delay constraint. For video streaming applications, there is a tradeoff between acceptable delay and tolerable packet loss. A delay constraint imposes an upper limit on this tradeoff since the lower the delay constraint, the greater the probability of packets being dropped due to exceeding the delay constraint.

Fig. B.14 shows the Complementary Cumulative Distribution Function (CCDF) of the FTD averaged over all content types with an increasing number of contending stations. For example consider a video streaming application with a DC of 500 ms, it can be seen that with 4 contending background stations, the FTD is always less than 500 ms. However with 6, 8, and 10 background contending stations, 2%, 12%, and 35% of video frames will have an FTD that exceeds a DC of 500 ms. The statistical distribution of the FTD has been summarized in Table B.5 which presents the CCDF of the FTD for different values of DC and with an increased number of contending stations.

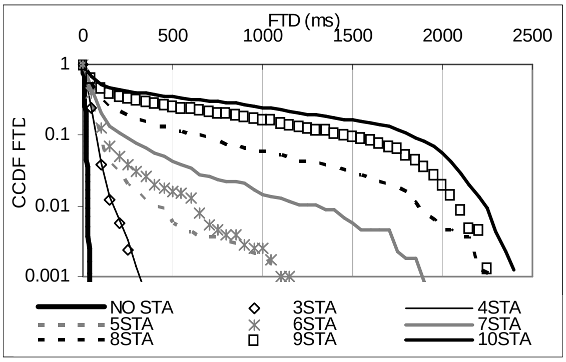


Fig. B.14: Generalized Distribution of The FTD with Increased Contention

Table B.5: CCDF of FTD Below the Playout Delay Constraint, DC

DC	Number of Contending Stations							
	3STA	4STA	5STA	6STA	7STA	8STA	9STA	10STA
500	1.000	1.000	0.994	0.984	0.957	0.877	0.740	0.653
1000	1.000	1.000	0.996	0.998	0.986	0.942	0.832	0.752
1500	1.000	1.000	1.000	1.000	0.994	0.971	0.903	0.836
2000	1.000	1.000	1.000	1.000	1.000	0.995	0.980	0.945
2500	1.000	1.000	1.000	1.000	1.000	1.000	1.000	1.000

It can be seen that when there are 10 contending stations, with a DC of 500 ms 65% of video frames will arrive within this upper delay bound whereas 95% of video frames will arrive within a DC of 2000 ms.

B.5.4 Conclusions

Here the effects of station contention on streaming video over IEEE 802.11b WLAN networks have been investigated. As the number of contending stations increases, while maintaining a constant total offered load, the video streaming application experiences increased delays. These delays are due to the IEEE 802.11b MAC mechanism where stations must contend for access to the medium. As the number of stations contending for access to the medium increases, the AP must defer decrementing the BC while another station is transmitting on the medium. Furthermore, it is evident that the complexity of the video content affects the degree of performance degradation.

B.6 Investigation of the Effects of *TXOP* on Parallel Multimedia Streams over QoS Enabled WLANs

The IEEE 802.11e standard defines four tuneable parameters namely – *ECW_{min}, ECW_{max}, AIFSN,* and *TXOP Limit* which can be used for prioritizing real time multimedia. We evaluate the performance of parallel multimedia streaming applications by providing differentiated service using the *TXOP Limit* parameter to under heavily loaded conditions. Various important factors like delay, loss rate, throughput etc. will be considered. We consider the performance of both the audio and video streams that comprise the multimedia session.

Video is bursty as each video frame is typically transmitted as a burst of packets. *TXOP Limit* is particularly suited to efficiently deal with this burstiness since it can be used to reserve bandwidth for the duration of the packet burst corresponding to a single video frame. We consider the delay required to transmit the entire video frame. The experimental results presented in this section were published in the International Conference on Communications (ICC) conference [25].

B.6.1 Dimensioning the *TXOP* Parameter

According to KIM *et.al.* [26] , the *TXOP Limit* parameter $TXOP_N$ is set to the number of packets required to transmit the video frame N_p multiplied by the time it takes to transmit each packet T_p during the *TXOP* interval.

$$TXOP_N = [N_p * T_p] \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad \text{(B.3)}$$

The time it takes to transmit a single video packet (T_p) during a *TXOP* interval is related to the packet size (P_{Sz}) and the physical line rate (*LineRate*).

$$T_p = \left(P_{Sz} / LineRate \right) + (2 * SIFS) + ACK \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad \text{(B.4)}$$

N_p *i*s the number of packets required to transmit the video frame of size F_{Sz} and is given by,

$$N_P = \left(F_{Sz} / P_{Sz} \right) \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad \text{(B.5)}$$

The distribution of the frame size is used to dimension the *TXOP Limit* parameter as it statistically describes the encoding characteristics of the video stream and the time required to transmit the video frame.

According to the IEEE 802.11e standard, the maximum allowable *TXOP Limit* is 8160 µs with a default value of 3008 µs. It is an integer value in the range (0,255) and gives the duration of the *TXOP* interval in units of 32 µs. If the calculated *TXOP* duration requested is not a factor of 32 µs, that value is rounded up to the next higher integer that is a factor of 32 µs.

B.6.2 Characteristics of Multimedia Streams

The properties of the five types of encoded clips were discussed in the 'tools' section B.2. This video content is approximately 10 minutes in duration and was encoded as MPEG-4 ASP (i.e. *I, P,* and *B* frames) with a frame rate of 24 fps, a specified refresh rate of 10 (i.e. an *I* frame every 10 frames), GOP sequence (i.e. *IPBBPBBPBB* resulting in 3 *I* Frames, 6 *P* frames, and 15 *B* frames per second), CIF resolution and a target bit rate of 1 Mbps using 2 pass encoding. In the experiments reported here the hint track MTU is 1024 B for all video content types.

Table B.6 shows the encoding characteristics of each of the different video streams and the average over all content types. It can be seen that high action and animation clips are particularly difficult for the encoder to achieve the target bit-rate.

Also, it can be seen that the combined load of the *I* and *P* frames is less than the load of the *B* frames only. This is due to the GOP structure of the video frames since there are on average three *I* frames, six *P* frames, and fifteen *B* frames per second.

Table B.6: Characteristics of the Video Content						
	DH	JR	EL	FM	DS	Mean Per Stream
Frame Rate (fps)	24.0	24.0	24.0	24.0	24.0	24.0
Mean Bitrate (kbps)	1633.0	980.0	1373.0	735.0	572.0	1058.6
Load *I* frames **(**kbps**)**	239.0	161.0	404.0	120.0	115.0	207.8
Load *P* frames **(**kbps**)**	407.0	315.0	457.0	202.0	170.0	310.2
Load *B* frames **(**kbps**)**	987.0	504.0	512.0	413.0	287.0	540.6
Mean/Max Frame Size (kb)	35.4	27.9	40.0	35.4	39.4	35.6
Mean/Max I Frame Size (kb)	53.7 /135.4	50.6 /103.7	109.3 /214.4	82.0 /139.9	69.3 /131.6	73.0 / 214.4
Mean/Max P Frame Size (kb)	18.6 /112.2	17.0 /89.9	37.1 / 130.1	27.5 / 130.2	23.3 /116.6	24.7 /130.2
Mean/Max B Frame Size (kb)	6.9 /200.8	6.4 /104.3	13.9 /112.3	10.3 /83.7	8.7 /92.4	9.2 /200.8
PMR	35.4	27.9	40.0	35.4	39.4	35.6

The audio content was encoded as MPEG-4 Advanced Audio Codec (AAC), 48 kHz, and 128 kbps CBR. The audio streams have the following characteristics: mean bit rate 130.93 kbps; mean sample size 341 byte; maximum sample size 667 byte; minimum sample size 52 byte; Peak-to-Mean Ratio (PMR) of 1.96.

B.6.3 Experimental Testbed

The video server and client were set up on the wired and wireless network respectively. Video streams were sent from server to the client via the Access Point. The AP used was the Cisco Aironet 1200 using the firmware version *IOS 12.3(8) JA* which allowed us to access the IEEE 802.11e/WME capability of the device. Differentiated Services Code Point (DSCP) value in the IP header was modified to force AP to send packets in different queues as the IEEE 802.11b default is to send all packets to best effort queue.

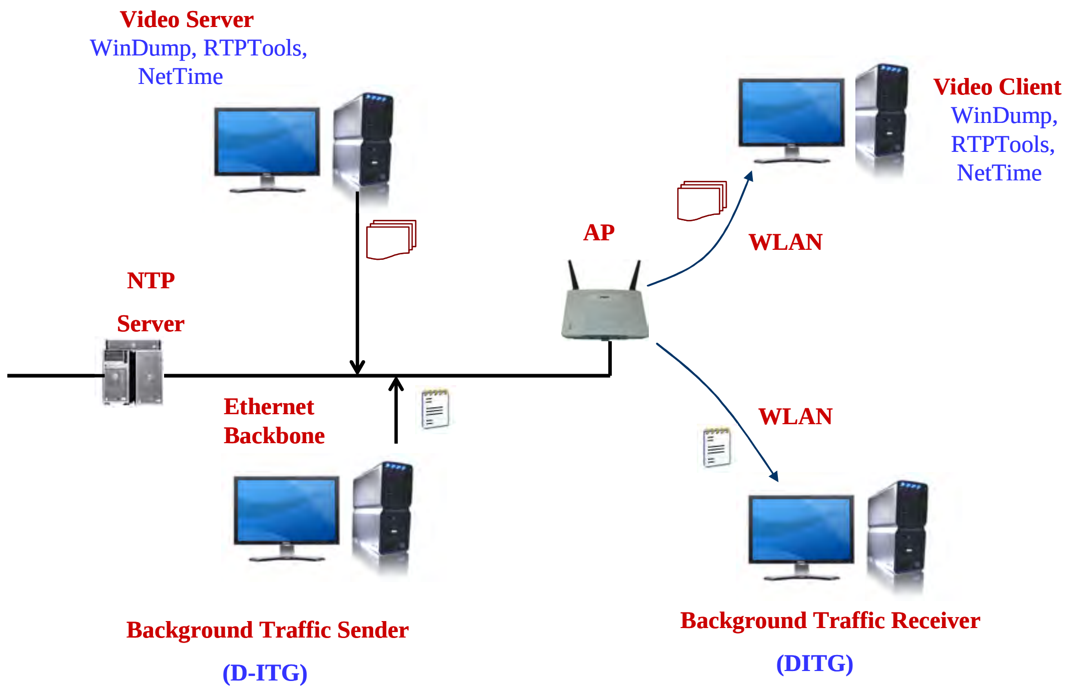


Fig. B.15: Experimental Testbed to Evaluate Effect of *TXOP*

The video streaming server consists of a modified version of RTPSender which reads from an encoded video file and identifies the different video frame types i.e. *I, P,* or *B* frames. The frame type indicator is used to set the DSCP value in the IP header. DSCP values are used to apply a particular Class of Service *(CoS)* to the incoming packets. Each *CoS* is then mapped to a particular AC where the *AIFSN*, ECW_{min}, ECW_{max}, *TXOP* parameters can be configured. In the experiments reported here, only the *TXOP Limit* parameter is varied and the parameters ECW_{min}, ECW_{max} *and AIFSN* were fixed with the original IEEE 802.11b settings. The default IEEE 802.11b values for ECW_{min}, ECW_{max}, and *AIFSN* values are 5, 10, and 2 respectively.

Windump, NetTime, and Paxson's algorithm were used for packet monitoring, clock synchronization, and skew removal in delay measurements respectively. Distributed Internet Traffic Generator (D-ITG) was used to generate background traffic, with characteristics of having an exponentially distributed inter-packet time with a mean offered load of 6 Mbps and an exponentially distributed packet size with a mean packet size of 1024 B. This traffic was transmitted from a wired source station via the AP to a wireless sink station.

B.6.4 Test Case Scenarios

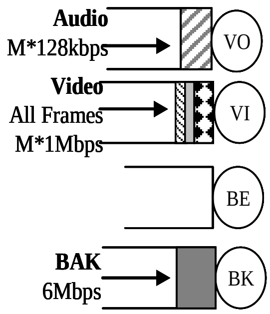
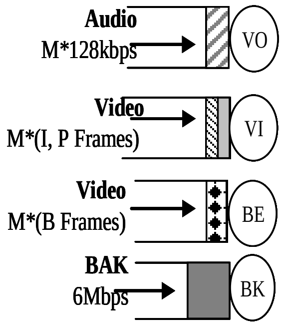
Description	Case	VO AC TXOP	VI AC TXOP	BE AC TXOP	BK AC TXOP
Default 802.11b	A	--	--	--	--
	B	0	0	0	0
	C	TXOP $(\bar{N}-\sigma)_{Audio}$	TXOP $(\bar{N}-\sigma)_{All}$	0	0
	D	TXOP $(\bar{N})_{Audio}$	TXOP $(\bar{N})_{All}$	0	0
	E	TXOP $(\bar{N}+\sigma)_{Audio}$	TXOP $(\bar{N}+\sigma)_{All}$	0	0
	F	0	0	0	0
	G	TXOP $(\bar{N}-\sigma)_{Audio}$	TXOP $(\bar{N}-\sigma)_{IP}$	TXOP $(\bar{N}-\sigma)_{B}$	0
	H	TXOP $(\bar{N})_{Audio}$	TXOP $(\bar{N})_{IP}$	TXOP $(\bar{N})_{B}$	0
	I	TXOP $(\bar{N}+\sigma)_{Audio}$	TXOP $(\bar{N}+\sigma)_{IP}$	TXOP $(\bar{N}+\sigma)_{B}$	0

Fig. B.16: Summary of Test Scenarios

Case A is used as a reference scenario where the AP uses the default IEEE 802.11b settings and all traffic streams are directed through a single queue. Since it was observed that the load from the *B* frames is approximately equal to the combined load of the *I* and *P* frames we investigate two key scenarios: where all video frames regardless of frame type are transmitted through the *VI* Access Category *(Cases B to E)* and where the *I* and *P* frames are transmitted through the *AC_VI* and the *B* frames are transmitted through the *AC_BE (Cases F to I).* In *Cases B to E* the audio, video and background traffic streams are transmitted through the *AC_VO*, *AC_VI*, and *AC_BK* queue respectively. In *Case B* the *TXOP Limit* parameter is set to 0. In *Case D* the *TXOP Limit* parameter is related to the mean number of packets $(\overline{N})$ required to transmit an audio sample i.e. $TXOP_{(\overline{N})AUDIO}$ and all video frames irrespective of frame type i.e. $TXOP_{(\overline{N})ALL}$. Similarly in *Cases C and E* the *TXOP Limit* parameter is related to the mean plus and minus one standard deviation of the number of packets i.e. $(\overline{N}+\sigma)$ and $(\overline{N}-\sigma)$ required to transmit the audio and video frames.

In *Cases F to I* the audio streams and background traffic are transmitted through the *AC_VO* and *AC_BK* queue respectively; the *I* and *P* video frames through the *AC_VI* queue and the *B* video frames through the *AC_BE* queue In *Case F* the *TXOP Limit* parameter is set to 0. In *Case H* the *TXOP Limit* parameter is related to the mean number of packets $(\overline{N})$ required to transmit an audio sample i.e. $TXOP_{(\overline{N})AUDIO}$, *I* and *P* video frames i.e. $TXOP_{(\overline{N})IP}$ and *B* video frames i.e. $TXOP_{(\overline{N})B}$. In Cases G and I the *TXOP Limit* parameter is related to the mean plus and minus one standard deviation of the number of packets i.e. $(\overline{N}+\sigma)$.

The number of multimedia streams was increased from 2 to 5 parallel streams for each of the different test cases. In all cases the *AC* queues were configured with IEEE 802.11b settings for ECW_{min}, ECW_{max}, and *AIFSN* while the value for *TXOP Limit* parameter is varied. Where used, the background traffic load was 6 Mbps in all cases.

B.6.5 Results

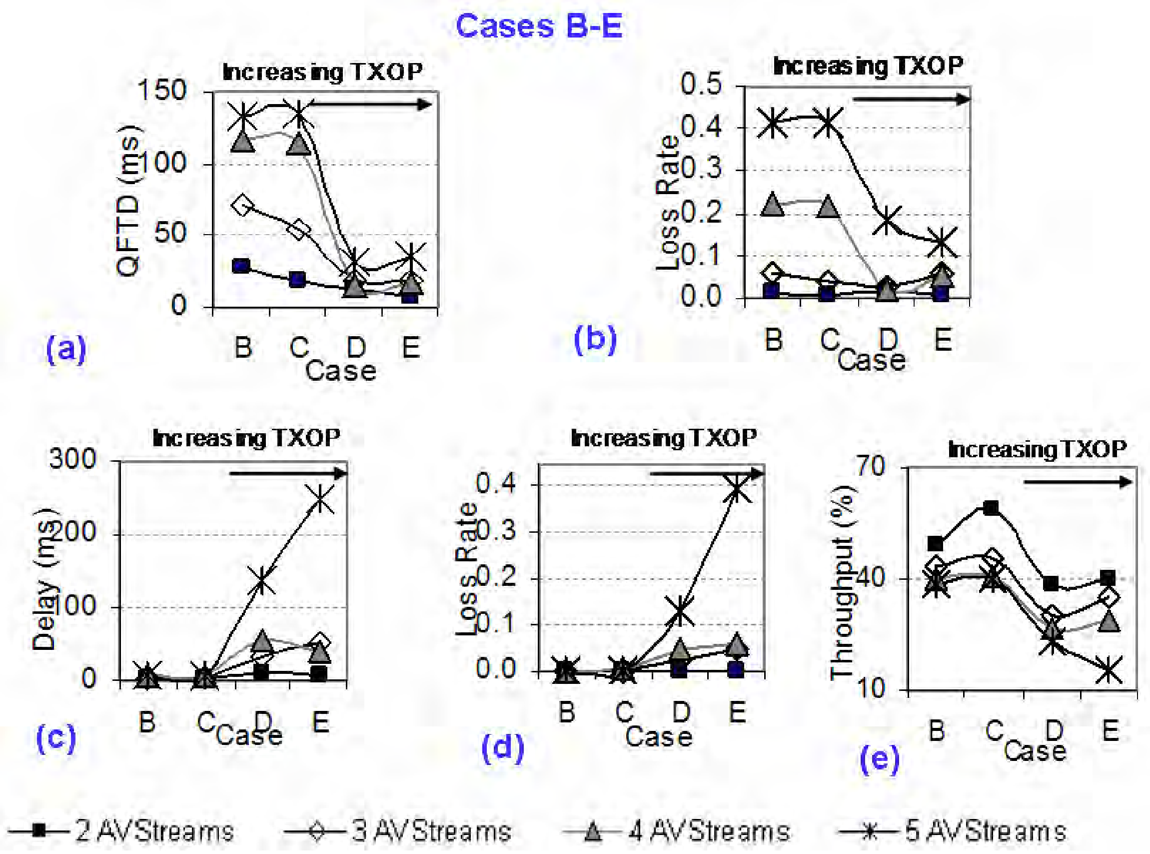


Fig. B.17: (a) Video FTD (b) Video Stream Loss Rate (c) Audio Sample Delay (d) Audio Stream Loss Rate (e) Background Traffic Percentage Throughput

Fig. B.17 shows the mean delay and loss measures with increasing values for the *TXOP Limit* parameter for *Cases B to E* with an offered background traffic load of 6 Mbps. Fig. B.17 (a) and Fig. B.17 (c) show the mean QFTD and packet delay for the video and audio streams respectively as the number of parallel streams is increased while Fig. B.17 (b) and Fig. B.17 (d) show the loss rates for the audio and video streams.

It can be seen in Fig. B.17 (a) and Fig. B.17 (b) that as the *TXOP Limit* parameter is increased for the video streams, the QFTD is reduced. The system can support 3 parallel video streams that satisfy a tolerable loss rate constraint of 5%. *Case D* exhibits the best performance having a QFTD of 18ms and loss rate of 3% for 3 parallel multimedia streams. Increasing the *TXOP Limit* parameter to the mean plus one standard deviation as in *Case E* increases the QFTD.

In contrast, it can be seen that the *TXOP Limit* parameter does not improve the end-to-end delay incurred transmitting audio samples. This is to be expected since an audio sample can be contained within a single packet and as such the *AC_VO* only needs to win a single transmission opportunity to transmit a complete audio sample. However by comparing the performance of the audio and video streams it can be clearly seen that as the *TXOP Limit* parameter of the *AC_VI* is increased, the performance (in terms of delay and loss) of the competing audio streams in the *AC_VO* deteriorates. This is particularly evident as the number of parallel multimedia streams is increased. This is due to the fact that usage of the *TXOP* is not wasteful since when the *AC* queue has won a *TXOP* and has no more packets to send during the *TXOP* interval, the HC senses the channel as idle and reclaims the channel after an interval of *PIFS* after the *TXOP*. As the number of video streams is increased the buffer occupancy of the *AC_VI* queue is also increased which in turn increases the likelihood that the *AC_VI* queue will make use of the full duration of the *TXOP* interval to transmit the queued video packets. Furthermore, as the *TXOP Limit* parameter of the *AC_VI* is increased it contends for access to the medium more often and as such gains access to the medium for longer intervals each time it wins a transmission opportunity. This in turn increases the waiting time for the *AC_VO* before it can contend for access to the medium thereby increasing the end-to-end delay for the audio samples.

Fig. B.18 shows the mean delay and loss measures of the audio and video streams with increasing values for the *TXOP Limit* parameter for *Cases F to I* with an offered background traffic load of 6Mbps. In this scenario the *I* and *P* frames of the video stream are transmitted through the *AC_VI* while the *B* frames are transmitted through the *AC_BE* queue. The results show that only 3 multimedia streams can be supported satisfying delay and loss constraints. It can be seen that as the *TXOP Limit* parameter is increased for the *AC_VI* and *AC_BE* queues both the *QFTD* and loss rate are significantly reduced.

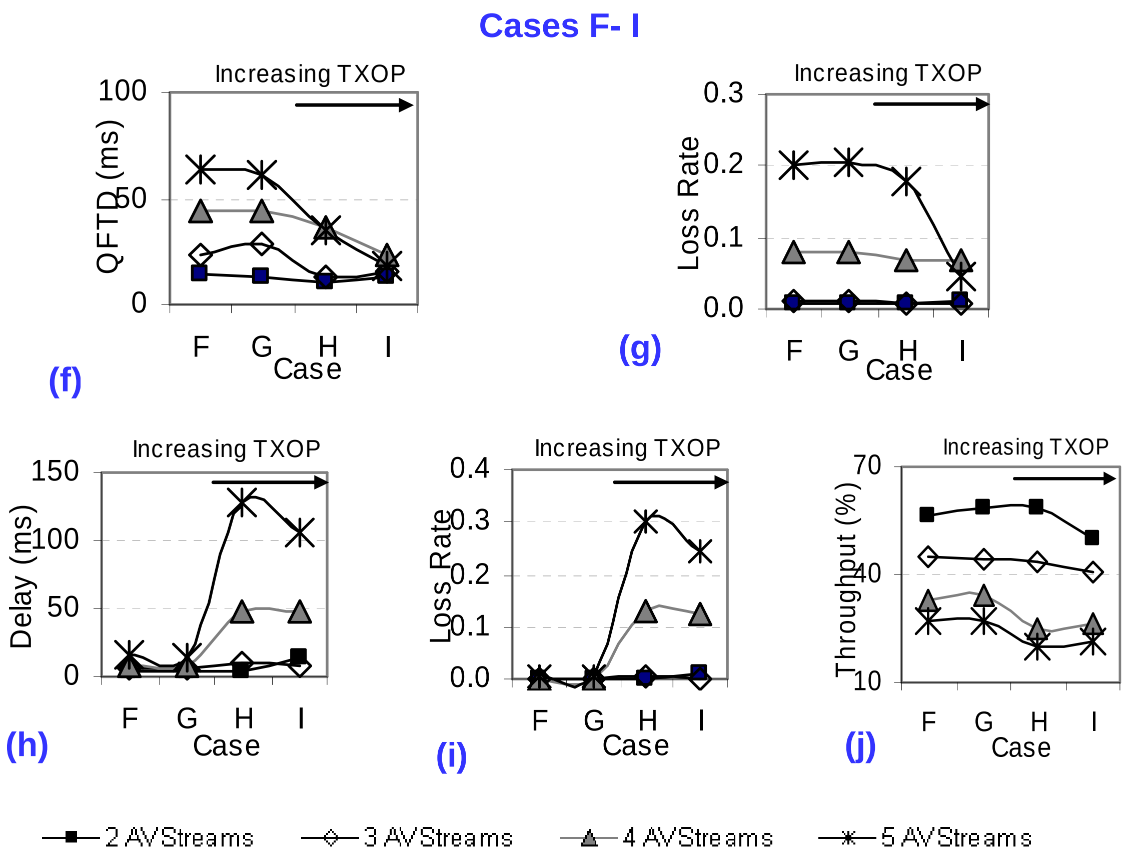


Fig. B.18: (f) Video QFTD, (g) Video Stream Loss Rate, (h) Audio Sample Delay, (i) Audio Stream Loss Rate, (j) Background Traffic Percentage Throughput.

By comparing Fig. B.17 and Fig. B.18 it can be seen that the performance of both the audio and video streams in terms of both the loss rate and delay is improved by transmitting the *I* and *P* frames of the video stream through the *AC_VI* and the *B* frames through the *AC_BE* queue as in *Cases F-I.*

Fig B.17 (e) and Fig. B.18 (j) show the percentage throughput of the background traffic for *Cases B-E* and *Cases F-I.* These figures show that the performance trade-off between the different *AC* using the *TXOP* facility becomes more pronounced in *Cases B-E* and that the throughput of the background traffic is greater in *Cases F-I*.

By contrast in *Case A* the AP is configured with the default IEEE 802.11b settings *(ECW_{min}, ECW_{max}, and AIFSN values* are 5, 10, and 2 respectively, No *TXOP* parameter present). When there is no background traffic the *QFTD* for the video streams increases from 9 ms with 2 parallel video streams to 26 ms with a loss rate of 5% for 5 parallel multimedia streams, i.e. 5 streams can be supported by the system. When 6 Mbps of background traffic is introduced, the AP becomes saturated resulting in buffer overflow. The throughput of the background traffic load is reduced to 73% while the video stream suffers a mean *QFTD* of 91 ms and loss rate of 59% which is unacceptable for multimedia streaming applications. In contrast when using IEEE 802.11e three multimedia sessions can be supported satisfying delay and loss rate constraints in the presence of 6 Mbps of background traffic using the *TXOP* facility. So by providing differentiated service to the different traffic streams in conjunction with the *TXOP* facility provides a significant performance improvement over the default IEEE 802.11b configuration.

B.6.6 Conclusions

Generally several packets are required to transmit a single video frame. The video frame cannot be decoded at the client until all the packets for the video frame have been received. But audio samples are transmitted at regular intervals and each audio sample can be contained within a single packet. *TXOP* does not improve the delivery of the audio samples but significantly improves the delivery of the video frame, i.e. *TXOP* is effective for reducing video streaming end-to-end delay, but not for audio streaming.

Over provisioning the *TXOP limit* parameter for the video streams in *AC_VI* access category causes the performance of the audio streams in the *AC_VO* access category to deteriorate and it becomes more pronounced as the number of multimedia streams increases. By providing differentiated service to the individual constituent *I, P, B* video frame types there is performance improvement for all ACs, we can reduce the likelihood of packets relating to *I* or *P* frames being lost since these frames have a higher priority and a greater impact on the end-user QoS over *B* frames. The rationale behind this is that the *I* frame should be given the best chance possible to be transmitted as the *P* frames and *B* frames are of little use without the reference *I* frame.

B.7 Summary of the Chapter

As discussed in earlier chapters, wireless links have low bit rate and high error rate along with time-varying bandwidth, delay, jitter, and loss characteristics. Real time traffic such as video cannot tolerate high bandwidth fluctuations hence guaranteed bandwidth and QoS are essential requirements for a high performance video streaming network. A wireless network designed to support high-quality video applications should provide for higher bandwidth and received bit rate with lower delay and jitter. In this chapter some insightful experimental results for video streaming on the IEEE 802.11b and IEEE 802.11e WLANs were presented.

Initially, the performance of a wired and wireless video server was investigated. Different characteristics of video streaming application namely video frame size, video frame rate, and packetisation scheme were varied. The impact on the received bit rate, loss rate, and end-to-end packet delay were reported. It was found that the received bit rate was much higher when using a wired server and large packetisation scheme. However, this can be traded off against an increased packet loss rate and increased delay when there are many concurrent streams. In contrast, the wireless server has a lower packet delay and loss rates.

Then the effect of background traffic on streamed video over IEEE 802.11b LANs for DLL and ULL in terms of loss, delay and bit rate was demonstrated. As the Down Link Load increases, the delay and loss rate of streamed video increase but received bit rate decreases. With increasing ULL, delay and loss rate increase slightly, but has no effect on received bit-rate.

Contention between stations was also analyzed. It was noted that as number of stations increases, contention negatively affects the QoD. Different video clips were affected in different ways. The animation clip was most severely affected on account of its greater spatial complexity.

The effect of transmission opportunity (*TXOP*) on streaming video and audio for the IEEE 802.11e networks was reported in section B.6. *TXOP* does not improve the delivery of the audio samples but significantly improves the delivery of the video frame, i.e. *TXOP* is effective for reducing video streaming delay, but not for audio. It was also found that over provisioning the *TXOP Limit* parameter for the video access category causes the performance of the audio stream to deteriorate and it becomes more pronounced as the number of multimedia streams increases. And the results indicate that by providing differentiated service to the individual constituent *I, P, B* video frame types and transmitting them via different ACs, there is a performance improvement for streamed video, i.e. video QoS is enhanced. These are quite important results as it is evident that the separate prioritization of constituent video frames (*I, P, B*) can be used for improving the quality of video that is transmitted over WLAN network. These results would be useful to design and dimension a video over WLAN system as they deal with the bandwidth and QoS issues directly.

Based on the experimental results described here, a unique QoS delivery algorithm is proposed, implemented and validated in chapter 4 and 5 which improves MPEG-4 video QoD over IEEE 802.11b WLANs by exploiting the *IPB* frame based nature of MPEG-4 video. The novel algorithm exploits the inherent coupling of two mechanisms, namely failed frame ReTx and GOPT to achieve the ITU-T target specified for loss rate ($\leq$1%) of streamed video transmission.

APPENDIX C – MAC Simulator Benchmarking Against NS3

Table (a): Throughput and loss rate comparisons for 1500 byte packets for 4 STA

1500 Byte Packets - 4 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.300	0.300	0.00	0.288	1.20
	50	0.600	0.513	8.74	0.574	2.60
	100	1.200	1.195	0.46	1.151	4.90
2	25	0.300	0.300	0.00	0.288	1.20
	50	0.600	0.529	7.09	0.575	2.50
	100	1.200	1.146	5.39	1.152	4.80
3	25	0.300	0.300	0.00	0.289	1.10
	50	0.600	0.525	7.52	0.576	2.40
	100	1.200	1.189	1.07	1.152	4.80
4	25	0.300	0.300	0.00	0.287	1.30
	50	0.600	0.544	5.64	0.577	2.30
	100	1.200	1.180	2.00	1.154	4.60

Table (b): Throughput and loss rate comparisons for 1500 byte packets for 3 STA

1500 Byte Packets - 3 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.300	0.300	0.00	0.289	1.10
	50	0.600	0.600	0.00	0.578	2.20
	100	1.200	1.200	0.00	1.155	4.50
2	25	0.300	0.300	0.00	0.288	1.20
	50	0.600	0.600	0.00	0.577	2.30
	100	1.200	1.093	10.70	1.154	4.60
3	25	0.300	0.300	0.00	0.288	1.20
	50	0.600	0.600	0.00	0.576	2.40
	100	1.200	1.090	11.04	1.153	4.70

Table (c): Throughput and loss rate comparisons for 1500 byte packets for 2 STA

1500 Byte Packets - 2 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.300	0.300	0.00	0.291	0.90
	50	0.600	0.216	38.4	0.582	1.80
	100	1.200	1.200	0.00	1.163	3.70
2	25	0.300	0.300	0.00	0.291	0.90
	50	0.600	0.216	38.4	0.582	1.80
	100	1.200	1.200	0.00	1.163	3.70

Table (d): Throughput and loss rate comparisons for 1024 byte packets for 4 STA

1024 Byte Packets - 4 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.210	0.210	0.00	0.197	1.30
	50	0.410	0.410	0.00	0.392	1.80
	100	0.820	0.722	9.80	0.786	3.40
2	25	0.210	0.210	0.00	0.196	1.40
	50	0.410	0.410	0.00	0.393	1.70
	100	0.820	0.702	11.81	0.786	3.40
3	25	0.210	0.210	0.00	0.197	1.30
	50	0.410	0.410	0.00	0.393	1.70
	100	0.820	0.696	12.39	0.787	3.30
4	25	0.210	0.210	0.00	0.196	1.40
	50	0.410	0.410	0.00	0.394	1.60
	100	0.820	0.694	12.63	0.788	3.20

Table (e): Throughput and loss rate comparisons for 1024 byte packets for 3 STA

1024 Byte Packets - 3 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.210	0.210	0.00	0.197	1.30
	50	0.410	0.410	0.00	0.395	1.50
	100	0.820	0.820	0.00	0.788	5.20
2	25	0.210	0.210	0.00	0.197	1.30
	50	0.410	0.410	0.00	0.394	1.60
	100	0.820	0.800	4.00	0.788	5.20
3	25	0.210	0.210	0.00	0.196	1.40
	50	0.410	0.410	0.18	0.395	1.50
	100	0.820	0.704	13.58	0.787	5.30

Table (f): Throughput and loss rate comparisons for 1024 byte packets for 2 STA

1024 Byte Packets - 2 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.210	0.210	0.00	0.199	1.10
	50	0.410	0.410	0.00	0.397	2.30
	100	0.820	0.820	0.00	0.794	4.60
2	25	0.210	0.210	0.00	0.199	1.10
	50	0.410	0.410	0.00	0.397	2.30
	100	0.820	0.820	0.00	0.794	4.60

Table (g): Throughput and loss rate comparisons for 512 byte packets for 4 STA

512 Byte Packets - 4 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.100	0.100	0.00	0.098	0.20
	50	0.210	0.210	0.00	0.196	1.40
	100	0.410	0.359	5.13	0.373	3.70
2	25	0.100	0.100	0.00	0.098	0.20
	50	0.210	0.210	0.00	0.196	1.40
	100	0.410	0.389	2.07	0.373	3.70
3	25	0.100	0.100	0.00	0.099	0.10
	50	0.210	0.210	0.00	0.197	1.30
	100	0.410	0.385	2.49	0.373	3.70
4	25	0.100	0.100	0.00	0.098	0.20
	50	0.210	0.210	0.00	0.197	1.30
	100	0.410	0.371	3.90	0.374	3.60

Table (h): Throughput and loss rate comparisons for 512 byte packets for 3 STA

512 Byte Packets - 3 STA						
STA	PPS	Offered Load	NS3		Our Simulator	
		Throughput (Mbps)	Throughput (Mbps)	Loss %	Throughput (Mbps)	Loss %
1	25	0.100	0.100	0.00	0.099	0.10
	50	0.210	0.210	0.00	0.187	2.30
	100	0.410	0.410	0.00	0.374	3.60
2	25	0.100	0.100	0.00	0.098	0.20
	50	0.210	0.210	0.00	0.187	2.30
	100	0.410	0.410	0.00	0.374	3.60
3	25	0.100	0.100	0.00	0.098	0.20
	50	0.210	0.210	0.00	0.187	2.30
	100	0.410	0.410	0.00	0.374	3.60

Table (i): Throughput and loss rate comparisons for 512 byte packets for 2 STA

512 Byte Packets - 2 STA						
STA	**PPS**	**Offered Load**	**NS3**		**Our Simulator**	
		Throughput (Mbps)	**Throughput (Mbps)**	**Loss %**	**Throughput (Mbps)**	**Loss %**
1	25	0.100	0.100	0.00	0.094	0.60
	50	0.210	0.210	0.00	0.189	2.10
	100	0.410	0.390	2.04	0.378	3.20
2	25	0.100	0.100	0.00	0.094	0.60
	50	0.210	0.210	0.00	0.189	2.10
	100	0.410	0.387	2.34	0.378	3.20

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