UNIVERSITY OF WITWATERSRAND

Performance of Wi-Fi Coordination Schemes for VoIP in the Presence of FTP Data.

Research report submitted in partial fulfillment of the requirements for the degree of Master of Science(50/50) in Engineering, Electrical

by

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9/22/2014

Declaration

I declare that this research report is my own unaided work except where otherwise acknowledged. It is being submitted for the degree of, Master of Science in Engineering, Electrical, at the Faculty of Engineering, University of the Witwatersrand. It has not been submitted before for any degree or examination at any other institution.

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Date: 22/09/2014

Signature:

Abstract

Evolved 3GPP cellular core networks have made co-existence of heterogeneous Wireless Access networks (HetNets) possible. The evolved core network along with the development of multimode end user devices have led to the realisation of converged Access Networks. Wireless Local Area Networks (WLANs) are assuming a prominent role in the telecommunications ecosystem due to their cost effectiveness, ease of deployment and operation in the free spectrum. Although WLANs are only data centric, there will be greater demand for Voice over Internet Protocol (VoIP) over WLANs as multimode smart-phones become accessible and operators integrate WLANs into their business models. Therefore, it is imperative that WLAN's ability to support VoIP services is thoroughly understood. Currently, the design of call admission control mechanisms for WLANs that support heterogeneous (data and voice) traffic is a challenging issue. The challenge stems from the difficulty of modelling the behaviour heterogeneous traffic, mixed VoIP and data traffic.

IEEE 802.11 WLANs use two types of medium access schemes, the polling based schemes and the contention based schemes. Both types of WLAN coordination schemes have not been thoroughly investigated for their ability to support VoIP over WLANs in the presence of File Transfer Protocol (FTP) data sessions. File Transfer Protocol (FTP) is a Transport Control Protocol(TCP) based file exchange protocol. TCP was optimised for wired networks and as a result it is unsuitable for wireless network. Furthermore, it was not optimised to co-exist with VoIP and as a result of its burstiness it has severe impact on the jitter, packet-loss and delay of VoIP traffic.

The purpose of the work presented in this report is to evaluate the performance of Distributed Coordinated Function (DCF), Point Coordination Function (PCF) and Enhanced Distributed Coordinated Function (EDCF) techniques' ability to manage Voice Over Internet Protocol (VoIP) over WLAN in the presence of contending heavy FTP data. The key question this work seeks to answer is, are the Medium Access Control (MAC) coordination techniques in their present form capable of carrying VoIP data in the presence of other data. In other words, how realistic is the deployment of VoIP services with FTP services in the same network, using the current coordination schemes for WLAN? Can these coordination schemes be improved by using current MAC enhancements such as fragmentation and increasing the Access Point buffer?

The study is carried out for IEEE 802.11g as this is still the most widely deployed standard. The performance is evaluated by setting up a network of stations that generate both voice and FTP traffic in OPNET. The two network configurations are 30-Voice stations and 30-FTP stations; 15-Voice stations and 45-FTP stations. Moreover, two codecs G.711 and G.723 are compared to assess the effect of codec selection on performance.

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Acronyms

AC-1	Access Category 1(Voice)
AC-2	Access Category 2 (Video)
AC-3	Access Category 3 Best Effort Data (DCF)
AC-4	Access Category 4(Background Data)
AMPS	Advanced Mobile Phone Systems
ANSI	American National Standards Institute
AIFS	Arbitrated Inter Frame Spacing
AP	Access Point
ATM	Asynchronous Transfer mode
BC-PQ	Back-off with Prioritized Queuing
BPSK	Binary Phase Shift Keying
BSS	Basic Service Set
BSC	Base Station Controller
BTS	Base Station Transceiver
CAPEX	Capital Expenditure
CODEC	Compression/Decompression
CDMA	Code Division Multiple Access
CFP	Contention Free Period
CP	Contention Period
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CTS	Clear To Send
CW	Contention Window
DAC	Digital to Analogue Conversion
D-AMPS	Digital Advanced Mobile Phone Systems
DSSS	Direct Sequence Spread Spectrum
DCF	Distributed Coordination Function
DIFS	Distributed Inter Frame Spacing
DTMF	Delivery Traffic Indication Map
EDCA	Enhanced Distributed Channel Access
EDCF	Enhanced Distributed Coordination Function
ETSI	European Telecommunication Standards Institute
FCS	Frame Check Sequence
GPRS	General Packet Radio Service
GGSN	Gateway GPRS Support Node
GSM	Groupe System Mobile
GUI	Graphical User Interface
HCCA	HCF Controlled Channel Access
HCF	Hybrid Coordinated Function
IEEE	Institute of Electrical & Electronics Engineers
IETF	Internet Engineering Task Force

IFFT	Inverse Fast Fourier Transform
IP	Internet Protocol
ISM	Industry, Scientific and medicine
ITU	International Telecommunications Union
MAC	Medium Access Control
MGCP	Media Gateway Control protocol
MSC	Mobile Switching Center
NMT	Nordic Mobile Telephone
NAV	Network Allocation Vector
OFDM	Orthogonal Frequency Division Multiplexing
OPEX	Operating Expenditure
OSI	Open System Interconnect
PCN	Personal Communications Network
PDC	Personal Digital Cellular
PCF	Point Coordination Function
PSQM	Perceptual Speech Quality Measure
PLCP	Physical Layer Convergence Protocol
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
QoS	Quality of Service
RAS	Registration Admission Status
RAN	Radio Access Network
RFC	Request For Comment
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
RTS	Ready To Send
SIP	Session Initiation Protocol
SDP	Session Description Protocol
SIFS	Short Inter Frame Spacing
SGSN	Serving GPRS Support Node
TACS	Total Access Communication Systems
TCP	Transport Control Protocol
TCP/IP	Transport Control Protocol over Internet Protocol
TS	Time slot
TXOP	Transmission Opportunity
UDP	User Datagram Protocol
VoIP	Voice over IP
Wi-Fi	Wireless Fidelity
WLAN	Wireless Local Area Networks
XOR	eXclusiveOR

1. Chapter 1: Background

1.1. Introduction

Wireless Local Area Networks (WLAN) and wireless cellular networks are steadily converging into a flat Internet Protocol (IP) architecture thereby creating a single multi-service network. The two technologies emerged out of the desire to access previously wired services wirelessly. The WLANs emerged out of the desire to access data/internet wirelessly, and as such the first generation of WLAN tried replicating the wired Ethernet Protocol. In contrast, cellular technology emerged from the desire to access voice telephony services while in motion [1].

In earlier years, evolution of wireless cellular networks was focussed on voice and data was viewed as an aside, whereas evolution of WLANs revolved around data and voice was an aside. These paradigms shifted in recent years towards data services as more data applications were developed and as it was becoming possible to provide voice as a data service [2]. This paradigm shift was propelled by the popularity of packet based speech, commonly known as Voice over Internet Protocol (VoIP) [2]. This technology allows transportation of both voice and data over the same underlying data network.

Cellular networks are unable to compete with the technologies such as Wireless Local Area Networks (WLANs) for filling in the capacity/coverage gaps at a lower cost, in order to create networks that can fulfil the following requirements [3]:

- Support a growing number of devices especially smart-phones.
- Provide efficient use of network resources, spectrum and backhaul infrastructure.
- Support always on and seamless mobility.
- Support real-time services seamlessly.

The focus of the work in this report is on the performance of MAC technologies for WLANs in particular for Wi-Fi/IEEE-802.11g [4]. The Medium Access Control (MAC) layer, also known as layer-2 in the Open Systems Interconnect (OSI) reference stack is responsible for managing access to the communications channel. The three Medium Access Control (MAC) schemes in Wi-Fi networks are Distributed Coordination Function (DCF), Point Coordination Function (PCF) and Enhanced-DCF [4].

The significance of the coordination/MAC schemes in Wi-Fi networks cannot be viewed in isolation, because Wi-Fi networks have assumed a critical role in the modern cellular centric telecommunications ecosystem. The role of Wi-Fi networks will be highlighted in order to put this research work into perspective.

In this chapter of the report we present a review of evolution of WLANs, evolution of Cellular networks, convergence of Cellular and WLAN networks, interoperability of cellular networks and WLANs and finally the relevance of these technologies in society. This review is a necessary backdrop and provides a context for the MAC issues that are investigated and presented in this report.

1.2. Background: Wi-Fi

1.2.1. Introduction.

Wireless Local Area Networks (WLANs) are short range wireless access systems. They were first formalised by the American regulator, Federal Communications Commission (FCC) in 1981 when the FCC opened the unlicensed spectrum in the 902-928 MHz, 2400 MHz and 5725-5875 MHz, later known as the Industrial, Scientific and Medical(ISM) [5]. The FCC also defined rules governing systems that can operate in this spectrum. Amongst these rules was that the systems have to be direct Sequence Spread Spectrum (DSSS) and Frequency Hopping systems with maximum transmit power of 1 Watt [5].

The WLANs were developed as a replacement for Ethernet based Wired Local Area Networks in offices to enable mobility and sharing of internet in office and campus environments. As a result the earlier WLANs were really derivatives of the Ethernet technology because they were intended to replace it [5].

They can operate in two modes known as the Ad-Hoc mode and infrastructure mode. In Ad-Hoc mode two devices that support the same WLAN standard can communicate over a direct path (Figure 1).

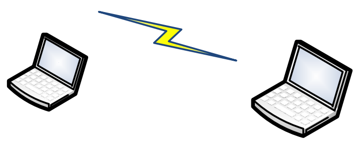


Figure 1: Wi-Fi Devices in Ad-hoc mode **[6]**.

In infrastructure mode there is a central relay point known as the Access Point(AP) that serves several devices (Figure 2). In infrastructure mode the same Access Point(AP) is shared by several stations over the air interface. The medium access control (MAC) sub-layer in the Link Control layer manages when each device including the Access Point can gain access to the medium.

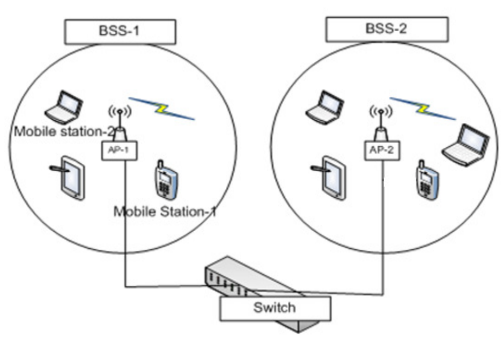


Figure 2: Wi-Fi Network in infrastructure Mode **[6].**

Unlike the wired channel, the wireless channel is hostile and prone to interference and losses, as a result the earlier WLANs suffered from slow speeds, expensive hardware, size and power consumption, as such they never really replaced wired LANs as was intended [5]. Consequently, their intended market which was corporate offices never really embraced them; hence they were relegated to vertical markets such as retailers, hospitals and logistics [5]. It was only in the early 1990s when the internet became popular that a real market for WLANs emerged, i.e. home networking, where the possibility of connecting home appliances that can share the network emerged. The shift away from enterprises to consumer began a revolution for this technology [5].

The earliest reported WLAN device was certified in 1988 [7]. At the time the devices were based on proprietary standards although later various standards such as HomeRF, IEEE 802.11, HIPERLAN emerged [5]. The key problem was that these standards were fragmented and therefore the devices could not inter-operate. This meant that consumers were locked to a manufacturer for all their home products. After years of rivalry and regulatory delays by the FCC, the IEEE 802.11 group of standards eventually emerged as a winning standard for WLANs.

The IEEE 802.11 group was formed in 1990 and was mandated to replicate the IEEE 802.3 Ethernet standards on the wireless networks [7]. This standard was finally completed in 1997 and it resulted in speeds of up to 2Mbps [5]. This standard allowed manufacturers too much flexibility. This resulted in many variant products based on the same standard but with interoperability issues, the exact issue that standardisation was seeking to address. As a result of problems with the standardisation of the 802.11 various manufacturers formed a consortium called Wireless LAN Interoperability Forum (WLIF) leading to the formation of a parallel standard called Home-RF. The purpose of the WLIF was to enable interoperability amongst devices from the various manufacturers such as Motorola, IBM and HP.

At around the same period the IEEE commissioned another group to investigate and standardise and improved version 802.11, with effective throughput of about 10Mbps. The resulting standard was called IEEE 802.11b, which was finalised and productised in 1999. Another parallel consortium called Wireless Ethernet Compatibility Alliance (WECA) was formed by rival manufacturers. This consortium punted the 802.11b standard under the pseudonym Wireless Fidelity (Wi-Fi) with a purpose of facilitating interoperability [5]. This consortium is today known as the Wi-Fi Alliance and the IEEE 802.11 standards are popularly known as Wi-Fi.

1.2.2. IEEE 802.11b

The IEEE 802.11b standard specifies operation in the Industrial, Scientific and Medical(ISM) frequency band, 2.4GHz [4]. In the 2.4GHz it is allocated 83.5 MHz bandwidth between the 2.4GHz and 2.48GHz, with channels that are 5MHz apart, as per Table 1 below [4]. The table shows the allocation in different regions.

Table 1: 2.4 GHz Regional Spectrum Allocation

Channel number	Central Frequency(GHz)	802.11 Allocation	Japan	North America	Europe
1	2.412	22 MHz channel			
2	2.417				
3	2.422				
4	2.427				
5	2.432				
6	2.437				
7	2.442				
8	2.447				
9	2.452				
10	2.457				
11	2.462				
12	2.467				
13	2.472				
14	2.484				

The IEEE 802.11b was the first true WLAN standard that experienced a large scale adoption. This standard specified a maximum 11 Mega bits per second (Mbps) bit rate with a 22 MHz channel in the 2.4GHz spectrum. This rate falls back to 5.5Mbps, 2Mbps and 1 Mbps as the channel radio conditions deteriorate [8]. The Table 1 above demonstrates that in order to have 22 MHz, a series of 5MHz channels have to be concatenated as in column 3. Therefore, in North America, channels 1, 6, 11 are the only three usable non-interfering channels of 22 MHz bandwidth [8].

The higher bit rates 11Mbps and 5.5Mbps are supported by the use of High Rate-Direct Sequence Spread Spectrum (HR-DSSS) as multiple access technique and the Quadrature Phase Shift Keying (QPSK) as the modulation depth on the radio interface. However, the lower bit rates 2Mbps and 1Mbps are supported by the DSSS as the multiple access technique and QPSK and BPSK modulation schemes, respectively [8].

In order to support different rates, both the transmitter and the receiver need to know the modulation scheme. The 802.11 standard appends a preamble known as the Physical Layer Convergence Protocol (PLCP) to the MAC frame that is destined for the receiver [4] to indicate the modulation scheme. Although the appending occurs at the MAC layer, the frame effectively relays the Physical layer information.

1.2.3. Home-RF ad HIPERLAN

The Home-RF standard [9] was designed to support data and streaming services [10]. This standard included Quality of Service support (QoS) which was a superior improvement on the 802.11 [10]. However, the major drawback of the Home-RF was that the products that were certified under this consortium were all based on Proxim's chipsets [5]. Therefore the industry viewed this as Proxim's attempt to hijack the Wi-Fi market. Finalisation of the IEEE 802.11b around the same time presented a threat to Home-RF, as a result Wireless LAN Interoperability Forum (WLIF) proposed an improved version to compete with the 802.11b. However, delays by the FCC in approving the Home-RF 2.0 standard, which it did in 2000, lead to uncertainty amongst members of the Home-RF/WLIF consortium and most of the members migrated to WECA thus spelling an end to Home-RF [5] [10].

In the USA, the WLAN industry and the market were given a freehand to decide the de facto standards. In contrast, the European governments under the European Telecommunication Standardisation Institute (ETSI) imposed the High Performance Radio Local Area Network(HIPERLAN) standard ETSI TR 101 683 [11], on the industry [5]. HIPERLAN-1 standard was approved in 1996. This standard was a far better performer than the incumbent IEEE 802.11b [5]. At around the same time the IEEE was standardising the IEEE 802.11a whose objective was to operate at minimum 20Mbps in the 5GHz band. The IEEE 802.11a standard adopted the HIPERLAN physical layer [5]. As a result of simpler implementation of the IEEE 802.11a and lack of support from USA regulators, the HIPERLAN standard never received large scale commercial support [5].

In 2002 the IEEE commenced a standardisation process to enable high speed throughput up to 20Mbps in the 2.4GHz band. This standard was required to be backward compatible with the IEEE 802.11b, the standard became known as the IEEE 802.11g. Three proposals were put forward for this standard, but only one of the proposals was in line with the regulation that allowed only direct sequence modulation schemes in the free band [5]. This regulation was later relaxed and the standardisation bodies were allowed to choose their standards based on performance. Ultimately, it was decided that the IEEE 802.11a standard will be adopted as is, in the 2.4GHz band, where it is now called the IEEE 802.11g [5] [4], with additional backward compatibility from 802.11g to 802.11b.

1.2.4. IEEE 802.11a/g

The IEEE 802.11a [12] and IEEE 802.11g [4] were the next evolution in the 802.11 standards following the 802.11b. They operate similar to IEEE 802.11b, but with one exception that they use Orthogonal Frequency Division Multiple Access (OFDM) as the multiple access technique instead of the HS-DSSS as was previously done for high throughput [4]. The 802.11a and 802.11g are similar; the main difference is that 802.11g operates in the 2.4GHz band whereas the 802.11a operates in the 5GHz band. The 5GHz is still largely unused at present and it has been proposed as reported in [13], that more spectrum should be allocated in this band to expand Wi-Fi services.

The 5GHz spectrum is not widely used, as a result it has 23 non interfering channels compared to the 3-non interfering channels in the 2.4GHz. The spectrum allocation is 3-bands of 100MHz each in 5.15-5.25 GHz, 5.25-5.35GHz, 5.725-5.825GHz [12]. This standard specifies maximum throughput of 54Mbps, with fall-back rates to 48, 36, 24, 18, 12, 11, 9, 6 Mbps over a 20MHz channel. The 6, 12 and 24Mbps are mandatory.

OFDM was introduced in order to overcome multipath fading, which is a problem in indoor wireless access networks [8]. By addressing multipath fading, which introduces inter symbol interference, OFDM has improved the theoretical data-rate by nearly five-fold, while using a narrower channel of bandwidth 20MHz. It works with 52 subcarrier frequencies, 48 user channels and 4 pilot channels. The subcarriers are numbered -26 to -1 and 1 to 26. The modulation depth can vary depending on the channel conditions as measured by the Signal to Noise ratio, with BPSK/QPSK used for lower rates and 16 QAM and 64QAM for higher data rates [12] see Table 2 below. In other words, in poor signal to noise environment, a more robust modulation scheme is used. The reader is referred to [14] for an in-depth analysis of the robustness of different modulation schemes.

Table 2: Modulation depths for IEEE 802.11 **[8]**

Data rate(Mbps)	Modulation Depth	
6	BPSK	Worst Channel Conditions
9	BPSK	
12	QPSK	
18	QPSK	
24	16-QAM	
36	16-QAM	
48	64-QAM	
54	64 QAM	Best Channel Conditions

1.2.5. IEEE 802.11n and IEEE 802.11ac

In 2004, the IEEE instituted another group to develop a higher throughput standard, with real data throughput of 100Mbps and raw throughput of 160Mbps as well as improved range. The standard was named IEEE 802.11n [4]. It was proposed that the standard will use a technique called Multiple Input Multiple Output(MIMO) and spatial multiplexing with channels of bandwidth 20MHz and 40MHz in order to improve the throughput. This has been widely accepted and implemented.

The next generation for IEEE 802.11 standards is the IEEE 802.11ac [4]. Compared to 802.11n, this standard improved the following aspects [15]:

- Support for wideband channels 20 MHz, 40 MHz, 80 MHz, 160MHz.
- 256 Quadrature Amplitude Modulation scheme.
- Multiple user MIMO.
- Beam forming.

It is expected that this standard will be finalised in 2014. The Evolution of the various WLAN standards is summarised in Figure 3.

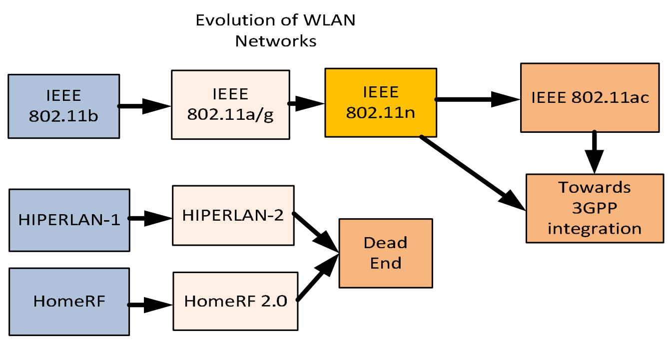


Figure 3: Evolution of WLAN Technologies.

1.2.6. Closing Remarks

The MAC layer of the Wi-Fi/IEEE 802.11 manages access to the medium in two ways, by using the distributed techniques and centrally coordinated techniques. The distributed techniques allow the device to contend equally with its neighbours for access to the medium. These schemes are called the Distributed Coordinated Function (DCF) and Enhanced-DCF. The centrally coordinated techniques allow the Access Point to control access to the medium; this is called the Point Coordinated Function (PCF) and Hybrid Control Function-Controlled Channel Access(HCCA). Both coordination mechanisms provide access to the medium while avoiding collision between data of different devices using a technique called Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). The technical details of these MAC coordination schemes are detailed in a dedicated chapter 2.

The evolution of WLANs from IEEE 802.11 to 802.11ac has focussed largely on improvements to the physical layer mechanisms (PHY). Other parallel standards were introduced along the way to improve the Medium Access (MAC) layer such as the IEEE 802.11e [4]. These MAC layer mechanisms were built to complement the existing Carrier Sense Multi Access with Collision Avoidance (CSMA/CA) MAC and they are ubiquitous across the main standards namely 802.11, 802.11b, 802.11a/g, 802.11n and 802.11ac, where the difference lies in the physical layer. Therefore, the behaviour of MAC layer mechanism can be studied under any of the standards and conclusions can be drawn about their behaviour in any other standards, as this is the essence of a layered architecture that separates functions and it is the founding principle of the Open System Interconnect (OSI).

1.3. Background: Cellular Networks

In modern telecommunication a discussion of Access Networks cannot be complete without touching on convergence. Furthermore, a discussion around convergence cannot be complete without a brief historical background of cellular networks. This section presents a brief history of cellular networks, from the first generation to LTE, in order to contextualise the convergence of cellular networks and WLANs.

1.3.1. First Generation Networks

The first generation of cellular systems, also referred to as 1G, appeared in the 1980s. At the time there was no global standard body for wireless cellular systems and as a result, these systems were fragmented across continents and international roaming was not possible [16]. Although, digital signalling was possible, data transfer was analogue and was based on Frequency Division Multiple Access (FDMA) [17], where each user is allocated a separate frequency to communicate.

Advanced Mobile Phone Systems (AMPS) was developed by Bell Labs, standardised by American National Standards Institute(ANSI) and was introduced in the USA in 1983 [18]. This standard was later adopted by China, Australia and South America. It operated in the 850 MHz frequency band, with 30 KHz full duplex channels, Frequency Shift Keying (FSK) modulation scheme and 45 MHz frequency spacing [18].

Total Access Communication Systems (TACS) was a variant of Advanced Mobile Phone Systems (AMPS) that was developed by Vodafone and adopted in the United Kingdom and Ireland [18]. Later Japan adopted the same standard and renamed it to JTACS [16] [18]. The notable difference between this standard and the original Advanced Mobile Phone Systems (AMPS) was that, in addition to the 800 MHz band, it also supported the 900 MHz band [16]. The channel bandwidth was reduced to 25 KHz as opposed to 30 KHz. The data signalling rate was also reduced.

Nordic Mobile Telephone (NMT) was first introduced in 1982 as a replacement for the 0th generation technologies Auto Radio Puhelin (ARP) meaning car phone, Mobile telephony system D(MTD) and *Offentlig Landmobil Telefoni(*OLT) meaning Public Land Mobile Telephony [16]. These technologies were used in Finland, Denmark and Sweden as well as Norway respectively [18]. It was developed for the 450 MHz band(NMT-450). It was later extended to 900 MHz(NMT-900) in order to increase capacity. The specifications of this technology were freely available, as such allowed many manufacturers to implement them. Other 1G system included the German C-450, the French RC 2000 and the Japanese NTT [18]. The Table 3 below summarises some key features of these standards.

The disadvantages of these systems were that analogue systems are spectrally inefficient, hence their capacity is limited. Analogue systems are vulnerable to noise, they are not secure, they were expensive hence hindered large scale uptake [16]. Furthermore, they were designed with voice service in mind and therefore were unsuitable for data services. Finally, international roaming was also not possible because the standardisation was fragmented. In order for devices to support these standards they would have to bulky and expensive, especially because the development of Silicon Integrated Circuits was not yet advanced at that time. These shortcomings lead to second generation systems (2G).

Table 3: 1G standards (adopted from **[16]**)

	AMPS	TACS	NMT (450/900)	NTT	C-450	RC-200
Uplink Frequency (MHz)	824-849	890-913	435-458/890-915	925-940	450-455.74	414.8-418
Downlink Frequency (MHz)	869-894	935-960	463-468/935-960	870-885	460-465.74	424.8-428
Modulation	FM	FM	FM	FM	FM	FM
Channel Spacing (KHz)	30	25	12/12.5	25	10	12.5
Number of Channels	832	1000	180/1999	600	573	256
Multiple Access	FDMA	FDMA	FDMA	FDMA	FDMA	FDMA
Signalling	Digital	Digital	Digital	Digital	Digital	Digital
Data	Analogue	Analogue	Analogue	Analogue	Analogue	Analogue

1.3.2. Second Generation Networks.

After a fragmented development of first generation systems, different countries started to appreciate the need for greater international collaboration in wireless communication system. This gave birth to the first international standardisation body in mobile telecommunications, Groupe System Mobile (GSM), later renamed Global Standard for Mobile communications. GSM became the de-facto standard mobile communications standard in most of Europe [16]. The standard was later adopted by many countries globally.

The American market persisted with a second generation Digital Advanced Mobile Phone Systems (D-AMPS) also known as IS-136 [18]. At around the same time, Qualcomm was testing a new Code Division Multiple Access (CDMA) system [16]. The CDMA system allowed users to share the spectrum via codes instead of Time Division Multiple Access or Frequency Division Multiple Access. It was standardized with a 1.25 MHz channel. The power of this system was highlighted by PacTel when they indicated that by using CDMA they could double the capacity of their analogue system with one tenth the spectrum [16].The first systems were commercialized under CDMAone/IS-95. Evidently, in the USA operators continued to adopt and deploy different standards, this continued to create difficulty with roaming and increased the costs of handsets as they needed multiple chipsets to support the different standards [16].

The commercial power of CDMA systems was demonstrated in Korea, wherein a number of subscribers reached 1-million in just months [18]. Other second generation systems such as Personal Communications Network (PCN) in the United Kingdom and Personal Digital Cellular (PDC) in Japan were developed but never reached global acceptance. GSM became the de-facto second generation global mobile communications standard because it was widely adopted in Europe and subsequently in most countries around the world. Therefore USA operators had to adopt GSM as their second generation standard [18], albeit without any obligation to discontinue the IS-95. The Table 4 below outlines some key differences between the various 2G standards

Table 4: Diffeence between 2G standards **[16].**

	GSM	IS-136(D-AMPS)	IS-95(CDMAone)	PDC
Uplink Frequencies(MHz)	890-915	824-849	824-849	810-830, 1429-1453
Downlink Frequencies(MHz)	935-960	869-894	869-894	940-960, 1477-1501
Channel Bandwidth(KHz)	200	30	1250	25
Modulation	GMSK	DQPSK	BPSK/QPSK	DQPSK
Multiple Access	FDMA/TDMA	TDMA	CDMA	TDMA
Channel Data rate	270	48	1.22 Mchips/s	42
Compressed Speech Rate	13	7.95	1.2-9.6	6.7

Later the 2G systems evolved to 2.5G systems in order to accommodate increasing data requirements. These systems included General Packet Radio Service (GPRS) and subsequently Enhanced Data rates for GSM Evolution(EGPRS) which enabled higher data rates**.** Other 2.5G systems such as Cellular Digital Packet Data(CDPD) which was data over D-AMPS network and Interim Standard-95B(IS-95B) emerged. The IS-95B was a CDMA based standard also known as the Narrowband CDMA (N-CDMA). This became the predecessor to the first third generation system known as the CDMA2000 [16]**.**

The key features of second generation networks (2G) were [16]:

- Digital data and signalling.
- Improved spectral efficiency.
- Improved security features on the air interface.
- Improved data rates.
  - GPRS with timeslot aggregation resulting in 56-114 Kbps.
  - EGPRS with variable modulation schemes and timeslot aggregation achieved 200-384Kpbs.
  - IS-95B with aggregate Walsh functions achieved 64-115 Kbps.
- Greater collaboration through the standard bodies, leading to International roaming.

The Figure 4 outlines some main features of the GSM network.

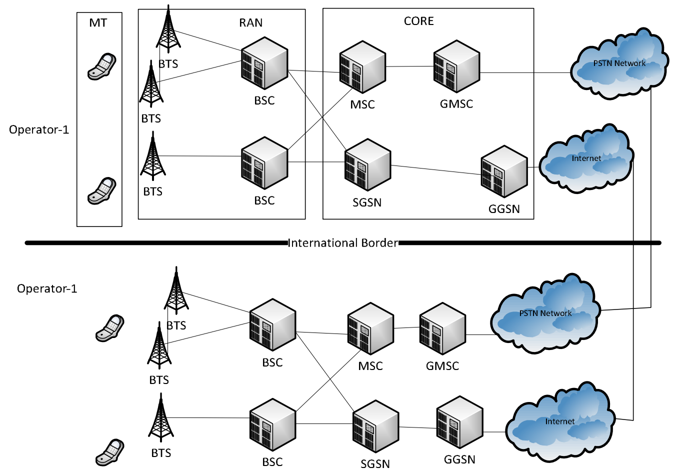


Figure 4: Elements of a GSM Network **[16].**

The GSM network was broken up into three domains namely, the Mobile Terminal (MT), the Radio Access Network (RAN) and the Core network. The core network consists of Mobile Switching Center (MSC), Serving GPRS Secure Node (SGSN), Gateway GPRS Node (GGSN) and the Gateway-MSC(G-MSC). The RAN is made up of the Base Station Transceiver (BTS) and Base Station Controller (BSC). Together the two elements facilitate the mobile terminal's access to the core network where its voice calls are digitally switched by the MSC and GMSC (for calls destined to other networks) and data packets are routed to and from the internet by the SGSN/GGSN. This segmentation became the basis for future evolution because it allowed the three domains to evolve and develop independently.

1.3.3. Third Generation Networks

The continued appetite for data services and persisting fragmentation of standards drove further evolution of 2G systems towards higher speed and a single global standard. As a result International Telecommunications Union proposed a future single global standard known as the International Mobile Telephone -2000 (IMT-2000) [16]. The proposed standard was proposed to provide high quality voice and high speed data services. CDMA-2000 which was preceded by CDMA-one and supported by Third Generation Partnership Project 2 (3GPP2) and Wideband CDMA (W-CDMA) supported by Third Generation Partnership Project (3GPP) which was an evolution from GSM, emerged as the two competing standards**.**

A consensus could not be reached over a single standard, North America opted for CDMA-2000, which was compatible with their already deployed CDMA-one and Europe and most of the world adopted WCDMA, which was compatible with the already deployed GSM [16]. The table below outlines some of the key differences between the two standards. The notable difference between these 3G systems and their 2G counterparts is that the channel bandwidth is a much larger, 5 MHz in W-CDMA compared with a 200 KHz channel in GSM systems. This misalignment once again created difficulties for mobile handset manufacturing because the handsets needed the chipsets to support both technologies because the two technologies were not compatible [16].

Table 5: Key difference between CDMA2000 and W-CDMA [16].

	CDMA2000	W-CDMA
Channel Bandwidth(MHz)	1.25 MHz	5 MHz
Chip Rate (Mchips/s)	1.2288 Mchips/s	3.84 Mchips/s
Peak Data rate (Mbps)	2.4	2.4(upto 8-10) with HSPA
Modulation Scheme	QPSK downlink & BPSK uplink	QPSK downlink & BPSK uplink
Coding Scheme	Convolutional(low rate), Turbo(high rate)	Convolutional(low rate), Turbo(high rate)

The W-CDMA third generation systems were standardised by the 3GPP and were compatible with GSM systems. They were designed to reuse the GSM core network elements (Figure-5). This ensured that operators did not have to reinvest in a new core network to support 3G systems, but with some software upgrades so that 2G core networks could support 3G services. The MSC server was introduced in order to support call control and mobility management aspects of the MSC [16]. However, new elements were introduced in the radio network in order to support 3G, namely, the Radio Network controller(RNC), the Node-B and the mobile terminal was renamed the User Equipment(UE) with support for both 2G and 3G air interface specifications. The RNC together with its Node-B's became known as the Universal Terrestrial Radio Access Network (UTRAN) and the 3GPP based 3G systems became generically known as the Universal Mobile Telecommunication System (UMTS) [16].

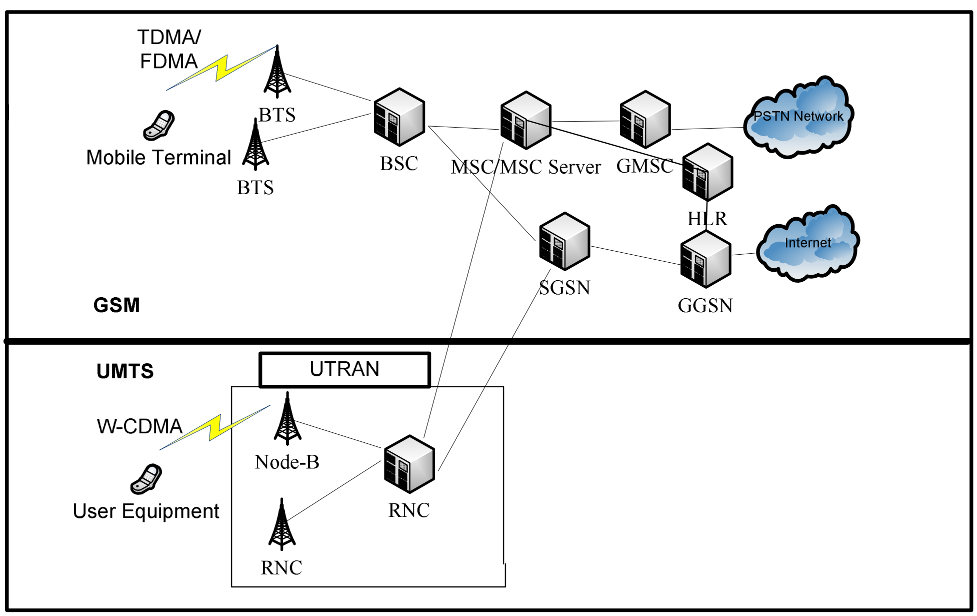


Figure 5: Diagram illustrating WCDMA reuse of GSM core network [16].

The W-CDMA based 3G systems were further improved with High Speed Packet Access (HSPA) and subsequently HSPA+. HSPA and HSPA+ introduced the concept of shared channel on the air interface, improved channel transition mechanism, improved error detection and correction mechanism on the medium access layer, higher order modulation schemes and carrier aggregation(dual carrier) in order to improve data rates [19]**.** These improvements are sometimes referred to as 3.5G.

CDMA2000 evolved to a standard referred to as EV-DO, Enhanced Voice Data Optimized. Although EV-DO maintained the 1.25 MHz channel bandwidth from CDMA2000, some improvements were made to the channels in order to achieve higher data rates.
Key features of 3G systems were:

- Improved spectral efficiency.
- Higher data throughput
- Improved latency.
- Backward compatibility with 2G systems.
- Improved collaboration towards defragmentation of standardisation bodies.
- Drive toward a flat all-IP architecture.

1.3.4. Beyond 3G

In 2008 International Telecommunications Union (ITU) created a framework for Fourth Generation systems under the auspices of International Mobile telecommunications–Advanced (IMT-Advanced), often referred to as 4G, to succeed IMT-2000(3G).
The ITU proposed that IMT-Advanced systems should amongst others have the following capabilities [20]:

- Provide a wide range of services supported by Internet Protocol (IP) packet based fixed and mobile networks.
- Support low latency and high mobility applications.
- Be able to tailor data rates to the user service for various user applications.
- The target transmission rate was set at 100 Mbps for mobile User Equipment and 1Gbps for stationary User Equipment.
- Support scalable bandwidth between 5MHz and 40MHz.
- Cell spectral efficiency: Uplink (UL) spectral efficiency of 3bits/s/Hz/Cell and Downlink(DL) 2.2bits/s/Hz/ cell.
- Peak spectral efficiency: UL 15 bits/s/Hz and DL 6.75/s/Hz with Multiple Input-Multiple Output(MIMO) 4 x 4 on the DL and 4 x 2 on the UL.
- User plane latency of maximum 10ms.
- Smooth call handover across heterogeneous(Het-GeNs) networks with maximum 60ms between frequencies.
- Voice over Internet Protocol, minimum 30 active users/sector/MHz.
- Worldwide roaming capability.

LTE-Advanced improved further on LTE in order to qualify as a 4G technology as per ITU-Advanced requirements. The following advancements were introduced in LTE advanced in order to meet the ITU-Advanced requirements [21] [22]:

- Carrier aggregation: This method allowed pooling of multiple carriers to carry data stream for a single user. A maximum of 5-carriers can be aggregated.
- Spatial multiplexing/MIMO.

- Relay Nodes: These are low power “repeaters” to enhance cell edge performance.

The changes to the core network in LTE systems especially the move towards all IP laid a foundation towards converged networks.

After a thorough assessment ITU accepted Long Term Evolution- Advanced (LTE-Advanced) from Third Generation Partnership Project (3GPP) and World Wide Interoperability for Microwave Access-Advanced(WiMax-Advanced) from the IEEE as meeting requirements for 4G. In this report only LTE is discussed.

1.3.5. Long Term Evolution (LTE)

In 2004 3GPP began work on the evolution of the 3G Universal Terrestrial Radio Access Network with a goal of developing “a framework for the evolution of the evolution of 3GPP radio access technology towards a high-data-rate, low latency and packet-optimized radio access technology” [23]. This framework became the basis for evolution of mobile telecommunication systems beyond 3G and towards 4G. The framework considered evolution of the network architecture and radio interface in order to deliver higher data rates, higher capacity and coverage at reduced cost. It was imperative that these requirements were met in order for the 3GPP standards to remain competitive against other competing standards from 3GPP2 such as CDMA2000 and IEEE such as the IEEE 802.16e (Mobile WiMax). As a result Long Term Evolution became the term that captures these objectives. This section will cover key aspects of LTE and System Architecture Evolution (SAE).

The 3GPP study group identified the following as the main study items towards LTE [23]:

- The Radio-interface was required to support scalable bandwidth from 5 to 20MHz, in steps of 5MHz. This is in order to enable data rates of 100Mbps downlink and 50 Mbps Uplink.
- Optimization of layer 2 and layer 3 signalling.
- Redesign of Universal Terrestrial Radio Access network (UTRAN) functional separation of the core network and the radio network.
- Support for packet service domain.
- Co-existence with other existing cellular technologies and bands (Inter Radio Access Technologies(RATs)).
- Packet based network but with support for real-time services, namely voice and video streaming and end to end quality of service.

The LTE architecture separates the Radio Access Network and the core network and refers to the two domains as the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) and an all IP Evolved Packet Core(EPC) respectively. The components in the radio network were further collapsed into to one entity, the eNode-B. Moreover, an additional interface was introduced between the Node-B’s. The UMTS abstraction of Access and Non-Access Strata, to separate the functions relating to radio access and those not related to radio access, was maintained.

The E-UTRAN became a collection of evolved-Node-B’s (eNode-B), providing user plane and control plane protocol termination towards the user equipment i.e. Layer-1 and Layer-2 services. The eNode-B’s communicate with each other within the E-UTRAN via the X2 interface and they communicate with the Evolved Packet Core via the S1 interface. The RNC from the 3G architecture was eliminated, as a result the eNode-B assumed some of the functions that were previously implemented by the RNC [21].

In order to increase the overall data throughput, the legacy W-CDMA multiple access scheme on the air interface was replaced with a new scheme called the Orthogonal Frequency Division Multiple Access(OFDMA) and a higher order modulation scheme 256 Quadrature Amplitude Modulation(256-QAM) was introduced [24]. The LTE system achieved higher data rates compared to previous generations due to a wider band of up to 20MHz, support for Spatial Multiplexing through Multiple Input Multiple Output (MIMO),

The functions of the LTE elements are summarised below and the architecture is highlighted in Figure 6**.**
In the highlighted LTE architecture, the functions of the eNodeB [21]:

- Radio Resources Management.
- Radio bearer control.
- Admission control.
- Scheduling.

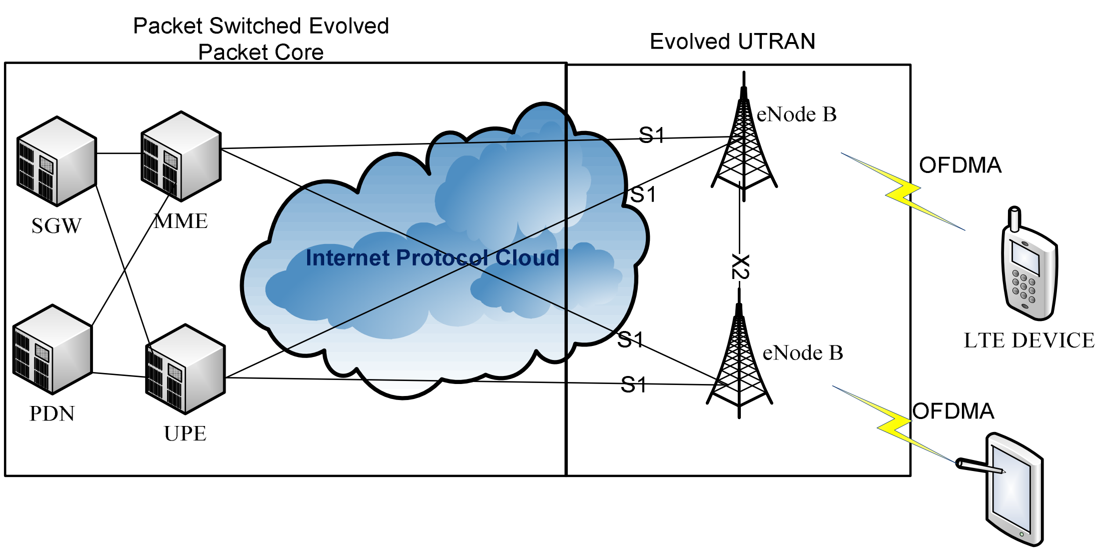


Figure 6: LTE Network layout **[25].**

The Evolved Packet Core comprises the Serving Gateway (SGW), Mobility Management Entity (MME), Packet Data Network Gateway(PDN-GW) [26].

The function of the Mobility Management Entity (MME) ([21]):

- Distributes paging messages to the eNode-B.
- Stores and manages UE contexts.
- Stores UE security parameters.
- User authentication.
- Roaming

Functions of the Serving Gateway (SGW), [21]:

- Conversion between SS7 signalling and IP based signalling for inter Radio Access technologies handover (Inter RAT).
- Mobility anchor for user plane during LTE handovers.
- Triggers paging for idle User Equipment, when new data arrives.

Functions of the Home Subscriber Server (HSS), [21]:

- The HSS is the master database.
- It contains subscriber related information.
- It provides support for mobility management.
- Provides support for Call and session setup.
- Provides support for user authentication and access authorisation.

Functions of the PDN-GW [21]:

- Interconnects the EPC to external networks and as such relays the IP packets to and from external networks.
- Performs Policy control and charging.

For detailed functions of these EPC elements the reader is referred to [21].

1.4. Converged Networks

The Second generation mobile systems were focussed largely on migration of voice services from analogue to digital, in order to improve spectral efficiency as well as a shift away from fragmentation towards global standardisation and international roaming. The second part of Second Generation introduced the idea of higher data speeds and for the first time made mobile internet a possibility.

Third Generation systems took the idea of mobile internet, up a gear. As a result of the popularity of mobile internet and the emergence of new services such as Voice Over Internet Protocol(VoIP), video streaming, interactive gaming, file sharing, social media and products such as mobile smartphones and tablets it has become clear that these cellular systems are unable to cope with the load. This lead to the introduction of Fourth Generation systems with much higher data speeds, to address the data hungry devices. Figure 7 summarises the different evolutionary paths that have led to convergence of HetGens over the Enhanced Packet Core(EPC) to provide seamless mobility for multimode devices, as shown in Figure 8.

The Evolution of cellular networks shifted the telecommunications paradigm away from circuit switched to packet switched services. The introduction of packet based services lead to development of new services and an unprecedented demand for high speed data access. The industry was trying to address the data demand through generational enhancements of cellular networks. In order to increase capacity allocation of additional bandwidth/spectrum is necessary, but the radio spectrum for cellular technologies is limited and expensive. Without additional spectrum, higher data rates cannot be achieved. Consequently, operators are forced to optimise their network development and consider alternative deployment architectures and technologies in order to meet the capacity demands [22]. An example of alternative architectures is deployment of short range small-cells. Wireless Local Area Networks (WLANs) have become an important aspect of the small-cell ecosystem because of the following reasons [22]:

- Most of the data hungry devices support Wireless Local Area Networks (WLANs).
- Wireless Local Area Networks (WLANs) networks are cheap and easy to deploy.
- Wireless Local Area Networks (WLANs) networks operate in the free Industry Scientific and Medical(ISM) spectrum.
- Wireless Local Area Networks (WLANs) standards are well defined and documented by the Institute of Electrical and Electronics Engineering (IEEE).
- The latest Wireless Local Area Networks (WLANs) standards lead to very high data throughputs.
- The Third Generation Partnership Group (3GPP) has developed standards to integrate Wireless Local Area Networks (WLANs) networks into the cellular network ecosystem.
- The Internet Engineering Task Force has developed protocols to allow mobility of Wireless Local Area Networks (WLANs) devices.

It is for these reasons that Wireless Local Area Networks (WLANs) has become a topic of interest in the cellular industry.

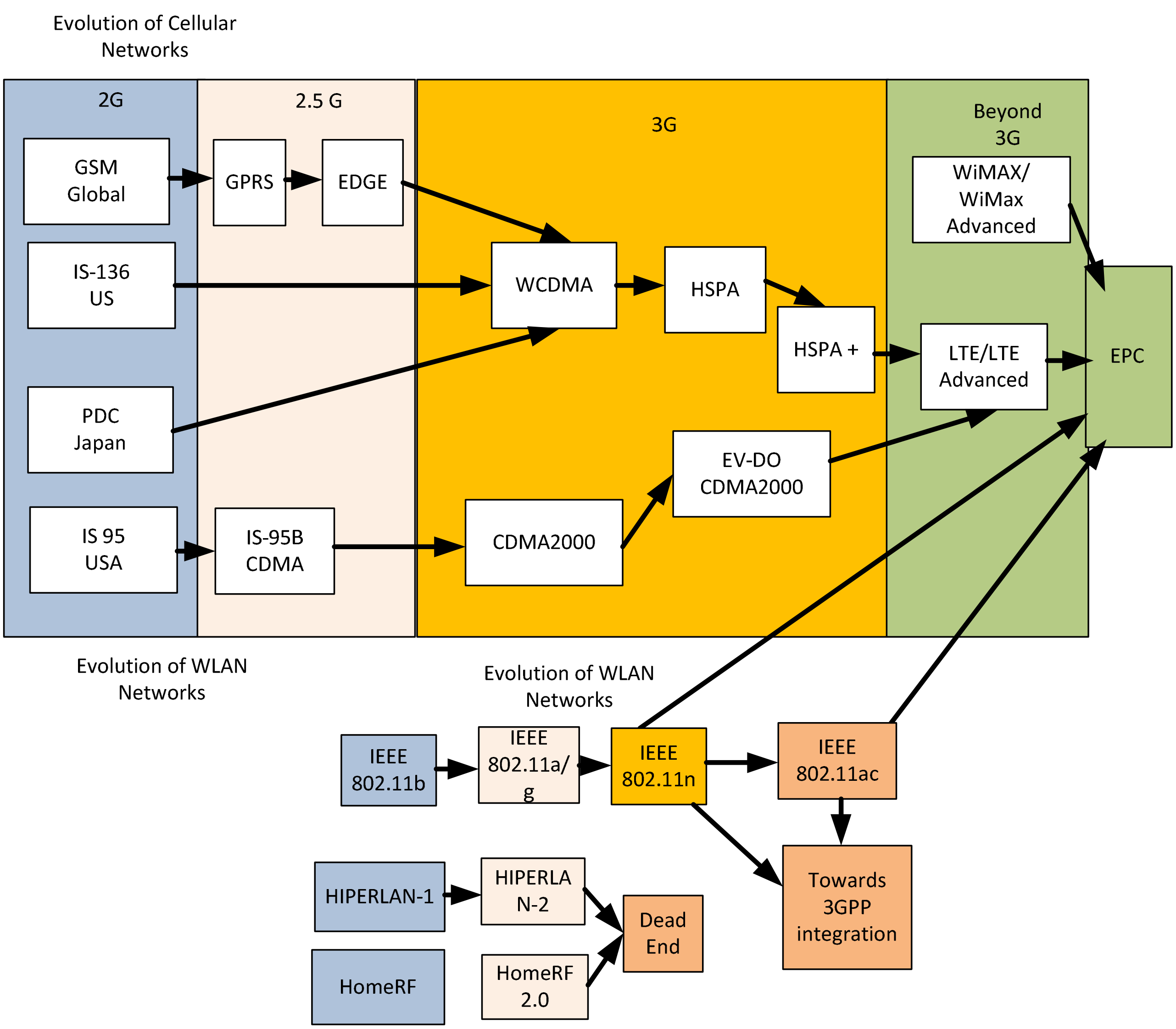


Figure 7: Evolution of cellular networks.

In light of the above review, the role of Wi-Fi as an offloading technology for the cellular networks, primarily 3GPP networks with an Enhanced Packet Core in order to support multimode devices (Figure 8) running multiple services across heterogeneous networks is discussed. The technical functions of layer-2 of

Wireless Local Area Networks (WLANs) networks are left to a dedicated chapter-2, as it is the main subject of this study.

Presently, most smart-phones and tablets support multiple services over separate networks that do not interoperate. Therefore, it makes sense that these networks are interoperable and can provide these services seamlessly. The interoperability would enable the end user to buy a converged service package from the operator, a single device and the user will always receive the best service from the best available access network without any manual intervention.

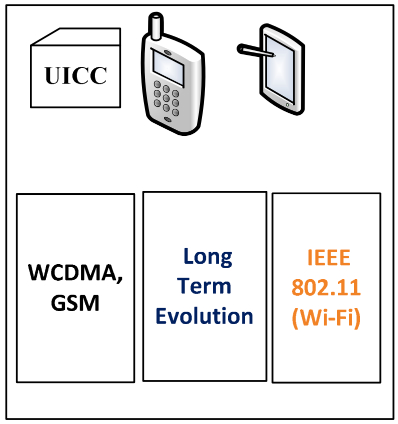


Figure 8: Multimode User Equipment

The technical functions of SAE are specified in 3GPP TS 23.402 in reference [27] and the high level architecture is demonstrated in Figure 9 below.

These are the SAE design objectives for interworking with non-3GPP technologies [27].

- Support access by non-3GPP access networks to 3GPP Enhanced Packet Core.
- Support Network based IP mobility management, which includes mechanisms to minimise handover latency due to authentication.
- Mobility management should avoid service interruption during handover i.e. enabling IP session continuity.
- Mobility management procedures should optimise User Equipment performance such as throughput and battery consumption.
- It supports interworking of 3GPP system with:
    - CDMA2000 based systems.
    - WiMax (IEEE 802.16).
    - Wireless Local Area Networks (WLAN).
- WLAN offload is supported both seamlessly and non-seamlessly.

In Figure 9 below the following elements can be identified in the Evolved Packet Core (EPC) the Mobility Management Entity(MME), Packet Gateway(PGW), Signalling Gateway(SGW), Authentication, Authorization and Accounting Server, Home Subscriber Server(HSS), Packet Data Gateway(PDG). The WLAN Access Gateway (WAG) can be placed either in the WLAN network or in the EPC core. The functions of these elements were described in the previous section on LTE. The functions of WAG and PDG will be covered shortly when the Wi-Fi/WLAN interoperability and offloading are discussed. Of significance is that the EPC has converged into a single core network supporting multiple heterogeneous networks, 2G, 3G, 4G and WLAN.

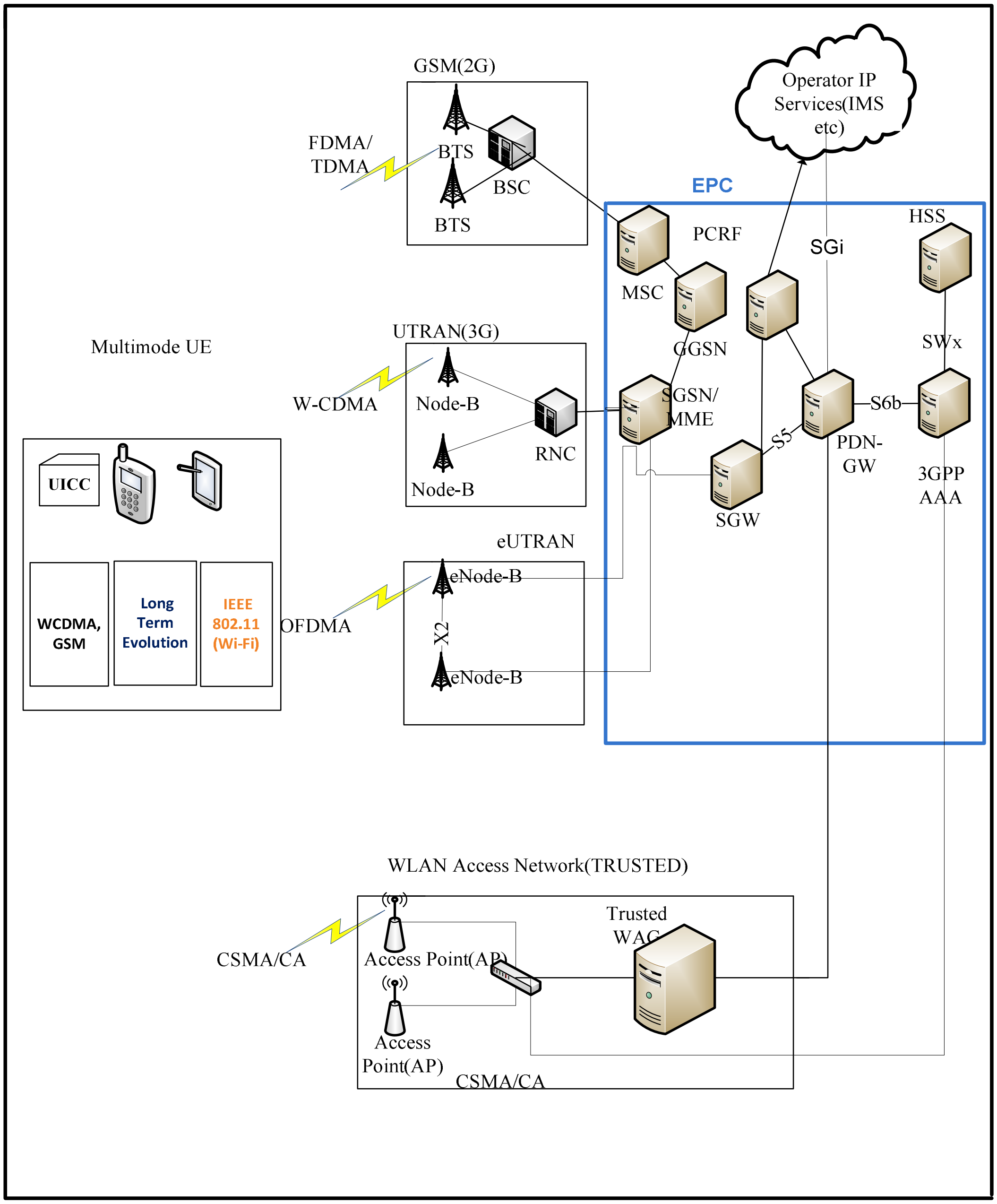


Figure 9: EPC Support for Multimode User Equipment across Heterogeneous Networks (HetGens)). [28]

1.4.1. Wireless Local Area Networks Interoperability with 3GPP Cellular Systems.

1.4.1.1. Interworking Architecture

The previous sections highlighted the supplementary role that WLANs have assumed in the modern telecommunications ecosystem. This section provides a detailed discussion on the interworking of WLANs and cellular networks. The main discussion points in this section are the architecture, access control and mobility (handovers) between cellular networks and WLANs.

The earlier UMTS based integration of WLAN was standardised under the 3GPP-I-WLAN with the following standards:

- 3GPP TS 23.234: Common Billing, Access Control and Charging and Access to PS Services.

- 3GPPTS 23.327: Service Continuity and Seamless Services – Single Radio Case and Mobility
- 3GPPTS 23.261: Service Continuity and Seamless Services – Dual Radio Case and Flow Mobility

These standards were extended to the Evolved Packet Core. For the purposes of discussing integration of WLAN and 3GPP only interoperability with the EPC is discussed in this report, the I-WLAN integration will not be discussed in this report and the reader is referred to 3GPP TS 23.234 [28], 3GPPTS 23.327 [29] and 3GPPTS 23.261 [30] for details of this integration. However, the EPC integration specification in [31]and [32] will be covered.

The requirement for WLAN-3GPP offloading was that, the WLAN should as far as possible reuse the Enhanced Packet Core(EPC) for as many services as possible and minimise the investment that operators have to make in order to deploy WLANs with the existing EPC [28]. The architectural framework is independent of the WLAN technology that is in use. Therefore, should an attractive WLAN standard emerge in the future and replace the IEEE 802.11 no new investment will be required in the Evolved Packet Core in order to integrate the new standard, refer to Figure 10 for the reference architecture.

In addition to the EPC elements outlined in the previous section on LTE the WLAN-3GPP offload requires additional elements namely:

- Trusted WLAN Access Gateway (T-WAG).
- 3GPP AAA Server.
- WLAN User Equipment (WLAN UE).

The WLAN Access Gateway performs the following functions [28]:

- Relays packet data between the WLAN access network and the 3GPP network, in order to provide 3GPP packet services to the User Equipment that is roaming in a WLAN network.
- It enforces relaying of data via the Packet data Gateway (PDG).
- It also gathers statistics for each tunnel for accounting purposes when a subscriber is roaming [28].

The 3GPP AAA server carries out the following functions [28]:

- Retrieves subscriber information from the subscriber’s HSS and authenticates the subscriber based on this information.
- Maintains the User Equipment attach status when the UE is attached to the WLAN.
- It provides suitable policy enforcement information to the WAG.
- When static routing is used, it provides the UE IP address as allocated by the HSS.

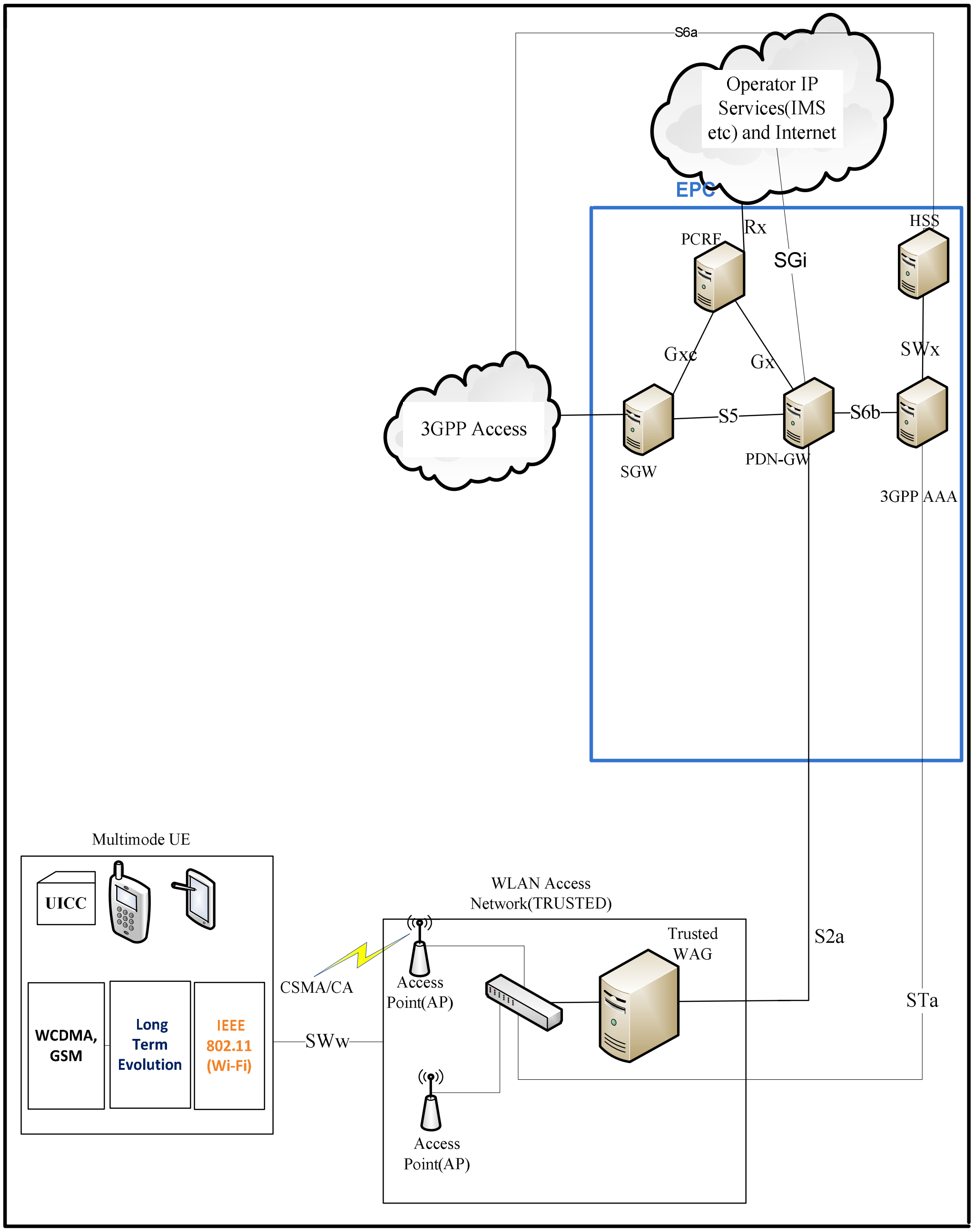


Figure 10: Trusted WLAN Integration with the EPC [28].

The specifications define the WLAN UE as a multimode device (supporting both WLAN and 3GPP networks) or a WLAN only device. This device must be equipped with the Universal Integrated Circuit Card (UICC) hardware. This is the hardware that enables support for Subscriber Identification Module (SIM), Universal-SIM (USIM) and IP Multimedia Subsystem SIM (ISIM) all on a single hardware device. As most modern devices support both cellular technologies and WLAN. The interoperability requirements proposed that management of the UE software should not be managed by operators, therefore all functions need to be handed by the network [28]. Furthermore, the requirements dictated that authentication methods have to rely on the current USIM or SIM authentication methods [28].

1.4.1.2. Access Control

Authentication methods are used to control the subscriber's access to the network. These authentication mechanisms should require no manual intervention from the subscriber in order for the subscriber to move between the different Radio Access Technologies (RATs) [28]. The authentication methods need to address two types of subscribers, operator subscribers with the Universal Subscriber Identification Module((U)SIM) cards and visiting subscribers with or without SIM cards [28]. WLAN networks use two types of authentication methods to address these needs, the Portal Based Authentication and the Extensible Authentication Protocol (EAP) [28].

The Portal Authentication methods address subscribers who do not have the operator's SIM cards such as those with vouchers and credit card payments [28]. This method relies on the Wireless Access Gateway (WAG) to authenticate subscribers. In other words, all the new subscribers that attach to the network are diverted to a portal which presents the user with a challenge to complete the Authentication Authorisation and Accounting (AAA) details [28]. If the user passes the AAA challenge the WAG allows the user's traffic to pass through the network [28]. Furthermore, the AAA server caches the device's hardware address for automatic re-authentication should the user disconnect from the network [28].

The Extensible Authentication Protocol (EAP) addresses the subscribers with the operator's SIM and as such allows subscribers to authenticate transparently (i.e. without manual intervention). EAP is specified by the IETF RFC 3748 [33]. The EAP only describes message format and flows, but the encapsulation of EAP messages is defined in IEEE 802.1X for all IEEE specified networks including the WLANs IEEE 802.11 [33]. The EAP methods allow presentation of different credentials for subscriber authentication. Examples of the EAP based authentication protocols include EAP-SIM [34], EAP-Authentication and Key Agreement (EAP-AKA, IETF RFC 4187) [35] and EAP-Internet Key Exchange version 2(EAP-IKEv2, IETF RFC 5106)) [36].

In cellular networks SIM (2G) and USIM(3G) card authentication details can be encapsulated in EAP messages for authentication and are referred to as EAP-SIM and EAP-AKA respectively [28]. When EAP is used the SIM/USIM subscriber authentication details are exchanged between the UE and the 3GPP AAA. The 3GPP AAA server relays these details to the HLR/HSS, see Figure 11 for the diagram. The typical authentication call flow is shown in Figure 11.

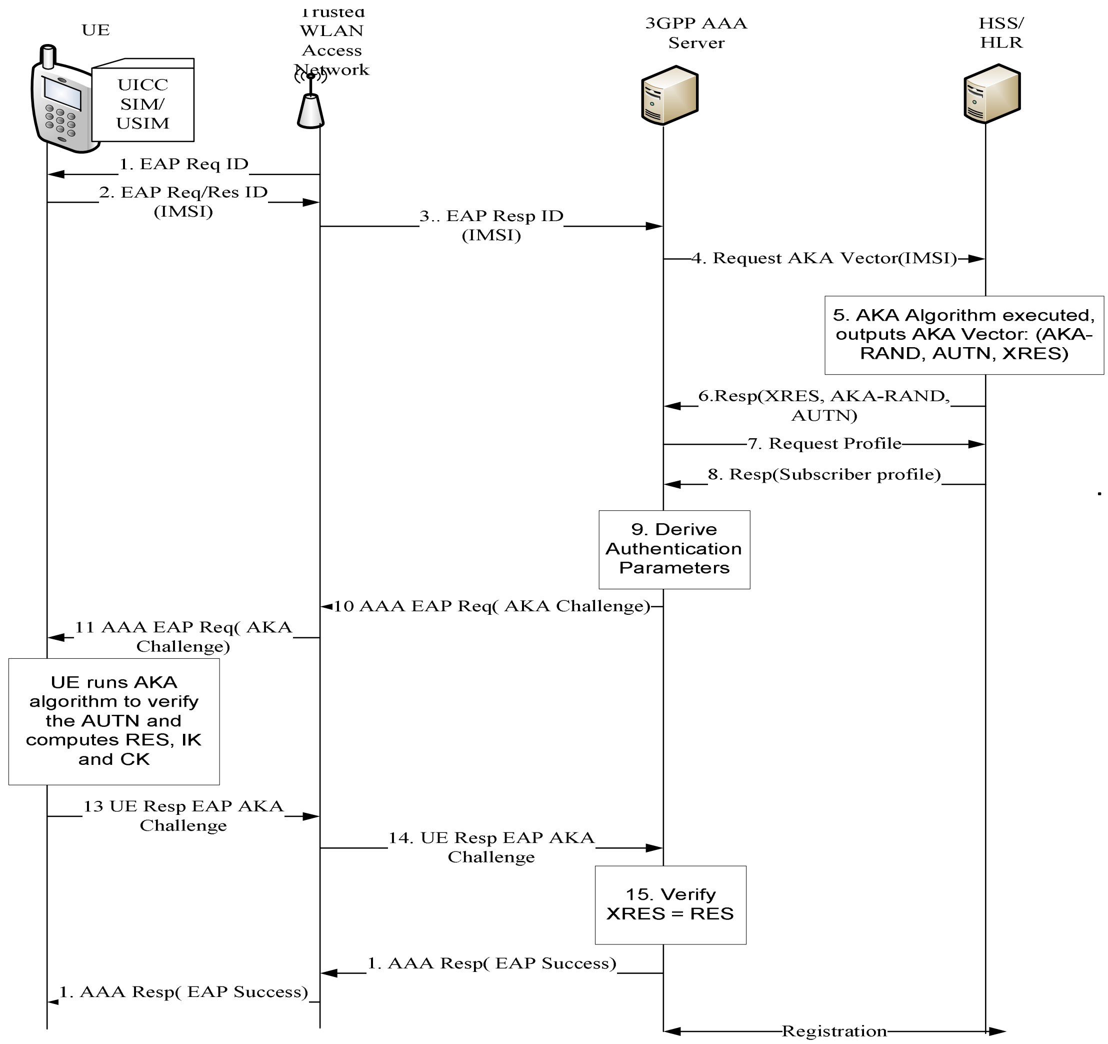


Figure 11: EAP-AKA Authentication call flow from **[37].**

EAP-AKA Method Call flow for USIM based Authentication [37]

The call flow can be summarised as follows:

1. The Trusted WLAN network (TWLAN) sends an EAP Request ID to the UE.
2. The UE responds with an ID, International Mobile Subscriber Identifier(IMSI)
3. The TWAN network forwards the Identifier to the 3GPP AAA Server, via the STa interface.
4. The 3GPP AAA server recognises the subscriber an EAP-AKA candidate and as such forwards the subscriber's ID to the HLR/HSS. If the Identifier vector does not indicate that the UE is capable of AKA, then the authentication procedure is terminated. If the UE support EAP-AKA, the 3GPP Server then check if there is an existing authentication vector for the UE's current Access Network, as indicated by the Network Identifier (NAI). If there is no existing vector the 3GPP AAA server then requests a new EAP-AKA authentication vector from the HSS.
5. The HSS runs an AKA algorithm based on the IMSI and returns the AKA-Rand, XRES and AUTN and responds to the request for AKA vector. The discussion of the authentication vector is outside the scope of this work.
6. The HSS returns the AKA-Rand, XRES and AUTN to the AAA server.
7. The 3GPP AAA server requests the subscriber profile from the HSS to verify that the subscriber is authorised to access the EPC.

8. The HSS responds with the subscriber's profile.
9. The 3GPP AAA server then computes the authentication parameters which include, Message Authentication Code (MAC), RAND and AUTH.
10. The parameters generated in 9 are then forwarded to the TWLAN access networks authenticator with an EAP challenge request.
11. The EAP challenge request is then forwarded to the UE.
12. The UE runs the AKA algorithm and confirms that the AUTN is indeed correct and there after the UE computes additional parameter the CK, IK and RES.
13. The UE sends the result of its AKA challenge, RES to the Access Network.
14. Access network forwards to the 3GPP AA Server.
15. The 3GPP AAA server then verifies that the UE generated RES = XRES.
16. The 3GPP AAA server then confirms EAP authentication and registers the UE on the HSS.

Once the UE is authenticated on the Trusted WLAN, the UE can perform network layer attachment and begin data sessions, see Figure 12 for the protocol stack. The network layer attachment details are presented in the next section along with the UE's handoff from the 3GPP Network (LTE) to the Trusted WLAN Network.

1.4.1.3. Handoff between WLAN and 3GPP Networks with EPC

Initially, integration of WLAN with cellular networks was done between traditionally cellular operators and traditionally Wi-Fi operators. As a result the WLANs were referred to as untrusted networks by the cellular community because they belonged to third parties [22]. As Wi-Fi gained traction with traditionally cellular operators, the cellular operators began building their own Wi-Fi networks [22]. The integration standards for the operator deployed WLAN networks came to be referred to as Trusted WLAN. As a result of these definitions different methods of integration were defined for the Trusted-WLANs (TWLANs) and Untrusted-WLAN. Integration of untrusted networks will not be discussed in this document.

In order for a UE to move seamlessly across heterogeneous networks an offload mechanism or IP handover mechanism is required because converged networks rely on All-IP architecture. There are two types of IP handoff mechanisms, namely the session handover with IP address persistency and the session handover without IP address persistency [22]. The session handover with IP address persistency refers to the type of handover wherein the User Equipment (UE) retains the same IP address while moving from a 3GPP radio network such as LTE to WLAN, whereas the non IP address persistent session handover refers to the opposite mechanism. The IP persistent mechanism is preferred in order to maintain an on-going session because IP traffic needs to be re-routed to the new access network from the core network.

The 3GPP WLAN interoperability employs three IP mobility management protocols namely the Proxy Mobile IP Protocol (PMIPv6), standardised in IETF's RFC 5213 in [38]; the Dual Stack Mobile IP(DSMIPv6) standardised in RFC 5555 and GPRS Tunneling Protocol (GTP). The DSMIPv6 protocol is an extension of the earlier version Mobile IP(MIP) RFC 5944, with support for IP version-6. The GTP protocol was initially standardised under by ETSI under GSM standard 09.60 and later migrated to the 3GPP under standard 3GPP TS 29.060 [39] and it is used both by the GPRS network and the UMTS network for data connection.
The key difference between these protocols is that the DSMIPv2 protocol manages the handover/offload procedure through the mobile device(UE) whereas with the GTP and PMIPv6 the handoff signalling

procedure is managed by the network and therefore does not require any active management from the device. The problem with the UE managed handoff procedure is that the operator has to also manage the software versions in the UE and this is a difficult task due to a large number handset/User Equipment make and models in the market. Therefore it is not favoured approach by operators and as a result this report will not discuss the DSMIPv6 based integration option any further.

The 3GPP TS 23.401 and 3GPP 23.402 define two integration options between the Trusted WLAN (TWLAN) and EPC, namely the S2c and the S2a. The S2c and S2a refer to the name of the interface that is in use between the WLAN network and the 3GPP network (see Figure 10). Essentially, the two interfaces terminate between the User Equipment (UE), Trusted WLAN Gateway (TWAG) and Packet Data Network Gateway (P-GW), but they are referred to by different names to distinguish the protocol that is in use, where the name S2c is used if IP mobility is managed by DSMIPv6 and the name S2a is used if IP mobility is managed by PMIPv6. The Untrusted WLAN is integrated via the GTP protocol and this is not covered in this report.
The Figure 12 below outlines the control plane and user plane protocol stack for the PMIPv6 through the S2a interface for the Trusted WLAN attachment. The control plane protocol stack is used to establish and tear down the PMIPv6 tunnels between the Trusted WLAN Access Gateway (TWAG) and the PDG-GW, the procedure for establishment of the PMIPv6 tunnel for both initial attachment and handover will be discussed in the next few sections. The User plane transfers user data and the control plane manages transfer of control plane data. Of significance is that the tunnelling layer terminates at the TWAG, this ensures that tunnel carrying user data can be established and terminated between the 3GPP core and the TWAG/another Radio Access Network without the mobile equipment being aware of this change. The mobile only has visibility of layer 4 in the user plane, in this layer the IP address of the mobile does not change hence this layer remains connected regardless of where the tunnel terminates.

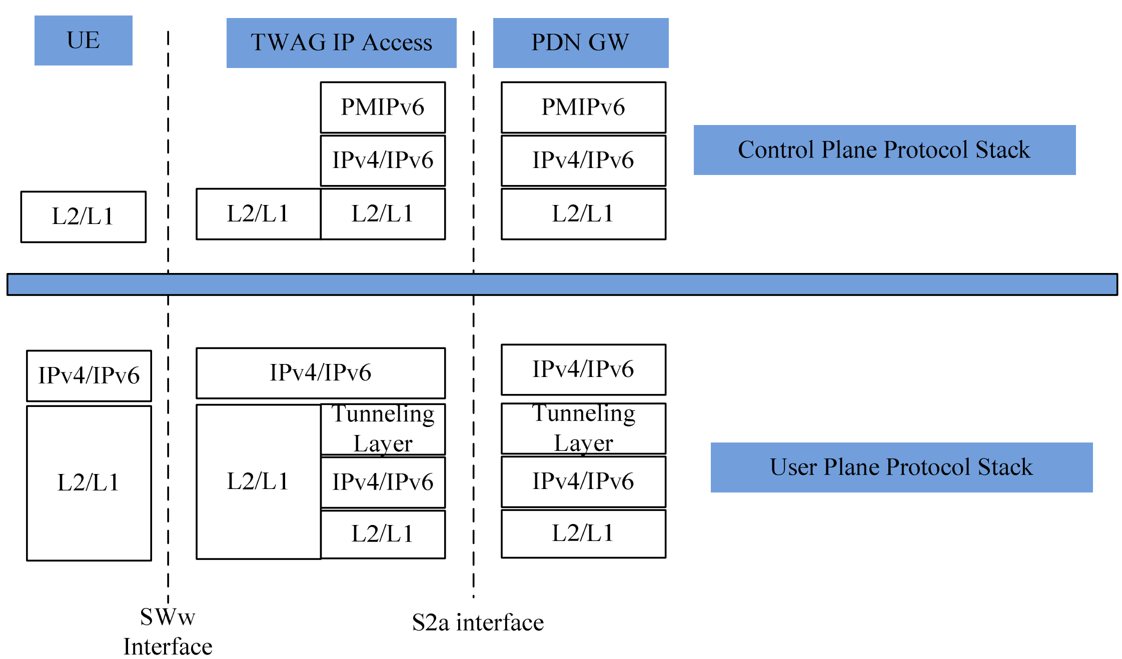


Figure 12: User plane and Control plane PMIPv6 Protocol Stack [27]

The PMIPv6 tunnel is established between the Trusted WLAN Access Gateway (TWAG) and the Packet Data Network Gateway in the Evolved Packet Core/Cellular core network. Then UE is assigned a cellular network

IP address, which comes from the same pool as all the other IP addresses that the cellular network assigns on the 3GPP access networks. The details of the call flow for the Network layer attachment procedure are outlined in the Figure 13 below and the protocol stack is outlined in Figure 12 above. Therefore in the user plane once the tunnelling layer is established between the cellular core and the TWAG, there is another IP connection that is established between the UE and the PDN. The tunnelling layer is indicated in the user plane protocol stack in Figure 12 as “layer-3”. Therefore, there are essentially two IP protocol stacks that run in parallel and are transparent to each other.

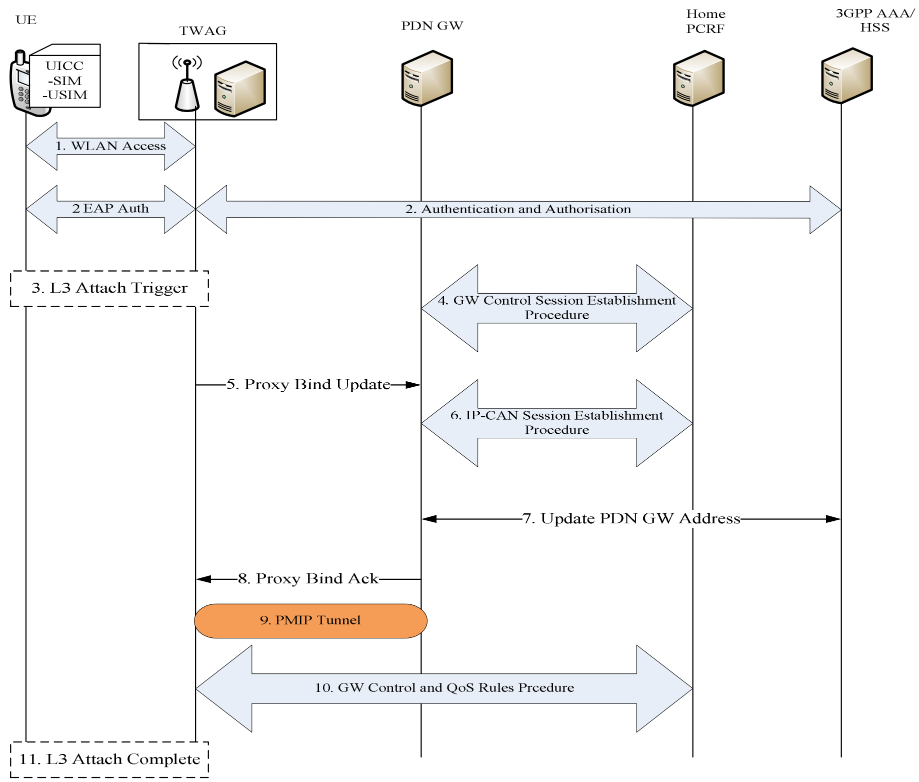


Figure 13: Layer-3 Initial Attach Call flow for PMIPv6 over S2a (3GPP TS 23.402) **[27]**

1.4.1.4. Handover Process

When a user is attached to the 3GPP network such as UMTS (3G) or LTE and the user moves into a WLAN network that has sufficient capacity to cater for the device, a handover process is initiated as outlined below, refer to Figure 14.

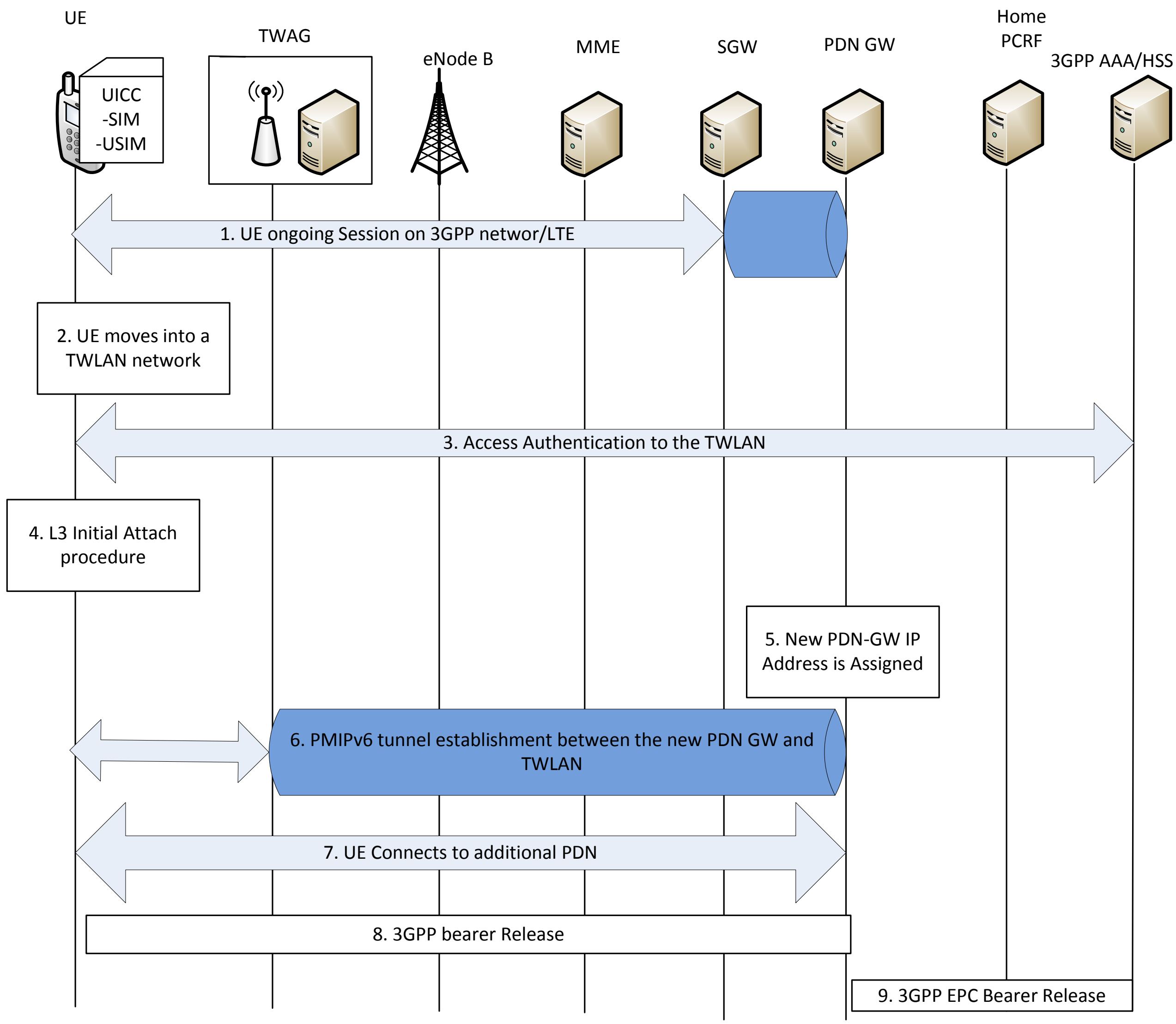


Figure 14: Call flow for the Handover process from LTE to TWLAN with PMIPv6 **[27].**

1. Initially the UE is connected to the 3GPP/LTE network and it is performing data services and therefore a GTP tunnel exists between the PDN-GW and the SGW.
2. The UE moves into the TWLAN network and kicks off a handover process.
3. TWLAN Access authentication procedure is carried out as in Figure 14 above(EAP-AKA Method Call flow for USIM based Authentication)
4. Once the UE is authenticated for the TWLAN network, a Network layer (Layer attachment) procedure begins. This procedure is the same as the initial attach as was outlined in L3 Initial Attach Call flow for PMIPv6 over S2a.
5. A new PDN GW may be used to serve the UE over the TWLAN network.
6. A new PMIPv6 tunnel is then established between the TWLAN Gateway and the serving PDN-GW.
7. In the case of persistent handover, there is no IP address change on the UE, therefore the UE continues to receive packets but it is unaware of the changes that have occurred at lower layer protocols.
8. The existing 3GPP tunnel is torn down and the 3GPP resources are released.

The most important feature of the PMIPv2 based networks is that the handover occurs without the UE's intervention.

1.5. Socio-Economic Relevance.

When investigating potential telecommunications technologies, it is always important to understand the socio-economic context of these technologies, because it is the socio-economic needs that determine how far technologies go. This section reviews the socio-economic status of VoIP over Wi-Fi.

According to CISCO systems in [40], the number of public Wi-Fi Hotspots is expected to reach 5.8 million globally by the year 2015, this translates to a 4 fold increase from the current levels. This, the report adds, is driven by the growth in handheld Wi-Fi enabled devices (smart phones and tablets) and the introduction of technologies that will simplify Cellular-to-Wi-Fi roaming experience. These seamless roaming technologies improve the ability of Wi-Fi devices to search and connect to the Wi-Fi Hotspot, securely. As a result of these developments, interest in Wi-Fi from operators has increased. In light of the increased interest, Taylor in [40] proposes several strategies to extract value from Wi-Fi:

- Wi-Fi can be an offloading network to preserve the limited spectrum shortage and also retain customers by providing increased quality of experience in indoor public spaces.
- Packaging Wi-Fi with cellular packages and allowing seamless roaming. The operators can sell Wi-Fi data at lower rates compared to cellular data. This would incentivise users to switch to Wi-Fi whenever possible, as such reliving the load off cellular networks.
- Using Wi-Fi to deliver Value Add Services (VAS) such as location based services, retail interaction, analytics and targeted advertising.
- Reselling access to Wi-Fi networks on wholesale basis. Operators can sell Wi-Fi access on wholesale to other vertical markets such as shopping malls, food chains etc. These companies can then package Wi-Fi as part of their offering.

Thomas Wehmeier of Informa Telecoms and Media in [41], cites several examples that highlight an increased interest in Wi-Fi from traditionally cellular and fixed-line operators, equipment manufacturers and start-up companies. Some examples that [41] highlights are activities by equipment manufacturers such as Ericsson's acquisition of BelAir Networks, Boingo's acquisition of Cloud Nine and Cisco's $1.2 Billion acquisition of Meraki. Richard Stuart of Meru Networks in [15] also claims that the predicted data explosion that accelerated investments in next generation high speed networks has arrived, but it is silently passing through Wi-Fi networks.

AT&T an American operator, pioneered inclusion of Wi-Fi into the data ecosystem through bundling of Wi-Fi data offerings with cellular data offerings on its 30000 deployed Hotspots. According to the same report [41] in 2011 AT &T reported 1.2 billion unique iPhone connections to the Wi-Fi networks. AT&T together with Apple were the first to implement Hotspot 2.0, this technology allows users to authenticate seamlessly when they are within Wi-Fi range. China Mobile positioned Wi-Fi as the core of its data offering and as a result, in the first quarter of 2012 it reported a total volume of 389TB data on its 2.3 million foot-print of Wi-Fi hot spots. However, the operator reports a far lower per megabyte revenue for Wi-Fi than on cellular network. KDDI, has deployed 220000 Hotspots and intends to offload 50% of their traffic by the end 2013. Boingo, a Wi-Fi only provider, reported 16.6 million unique connections in 2012.

In the local South African market, both Vodacom and Telkom have deployed Wi-Fi offload as part of their broadband strategies [42] [43].

The driving force behind Wi-Fi according to the report is the proliferation of smart-phones in the developed markets. The Table 6 below highlights how users in all the markets where data is available rely heavily on Wi-Fi networks. The table does not provide a breakdown of where the Hotspots are located. In most cases these are located in public places where users may be on the move and using Hotspots to pass time such as in airports lounges and restaurants.

Table 6: Distribution of Wi-Fi and Cellular subscribers in different markets **[41]**

Distribution(%)		
Country	Cellular	Wi-Fi
Thailand	27.9	72.1
France	20.6	79.4
Brazil	19.2	80.8
USA	31.6	68.4
Canada	22.9	77.1
Italy	29.2	70.8
India	47	53
UK	18.3	81.7
Hong Kong	27.5	72.5

In developed markets, an average iPhone user consumes 4GB of data per month and 3GB for Android users, of this data 82% and 66% respectively, is consumed over Wi-Fi [41]. A recent announcement by Whatsapp, in [44], that it will be launching the voice service is a further sign of diversion away from circuit switched voice. Whatsapp is the largest instant messaging service with an estimated 500 million users, as such for the first time a global user base will be exposed to Voice over IP. Some of the local South African operators have announced their intentions to charge a premium for data that is used to connect voice calls (VoIP) and have filed their intentions with the regulator [45]. It is the opinion of this report that operators are within their rights to charge a premium for this service; however, this may not be the best solution to compete with an alternative technology. Therefore, there is a case for Wi-Fi bundling as part of cellular offerings in order to offload traffic from the cellular networks.

In [13] a European Commission (EC) study in 2012, reported that 71% of all European Union (EU) smart-phone and tablet data traffic in 2012 was delivered over Wi-Fi. This figure is expected to increase to 78% by 2016. The EU further proposed that additional spectrum in the 5150 MHz to 5925 MHz should be allocated, globally, for Wi-Fi in order to cater for the smart-phone traffic. The migration of smart-phone users to Wi-Fi networks and the availability of VoIP applications such as Viber, Facebook Messenger, Whatsapp and Skype has created a fertile ground for growth in VoIP services over Wi-Fi networks. It is the author's view that cheaper smart-phones from Chinese manufacturers will lead to an inevitable smart-phone era, where smart-phones will become the norm. Moreover, as user's sophistication with smart-phones grows and the cost of acquiring smart-phones drops, more uses will replace their legacy handsets with smart-phones. Hence, it can be expected that users will migrate their voice services away from the traditional circuit switched networks towards data networks, where the cost per minute is relatively cheaper.

Chetty et al, in [46], proposed a VoIP over Wi-Fi as a cheaper solution for rural connectivity. They reported that in a Dominican Republic village, a Very Small Aperture Terminal/Wi-Fi solution was deployed within 3 days. Similar projects were trialled in China, India and South Africa with encouraging results. In the South African demonstration, the service was used to provide connectivity to public services such as schools, police stations, hospitals and clinics. Therefore, the potential value of VoIP over Wi-Fi in enhancing rural connectivity cannot be over stated. This service can bring remote villages closer to the rest of the world and in the process enable delivery of world class education, better health care and other government services such as social security, identification, drivers' licensing etc.

Based on above background, it is clear that Services over Wi-Fi, especially VoIP will become a prominent part of the telecommunications ecosystems and Wi-Fi traffic growth will continue for years to come. In addition it may also potentially solve pitfalls with rural connectivity by introducing a low cost connectivity solution for servicing rural areas, which are typically not revenue generating but where coverage may be part of regulatory obligations.

1.6. Problem Statement

1.6.1. Background

It is clear from the discussions above that the improvements from one IEEE 802.11 standard between generations has focussed primarily on the physical layer enhancement. The original Medium Access (MAC) layer mechanisms that were adopted from HIPERLAN-2 remained largely unchanged. These MAC mechanisms are now required to support real-time services together with disruptions from other traffic types such as File Transfer Protocol traffic (FTP) on the same Access Point, in spite of the fact that Wi-Fi networks were not designed with these requirements in mind. The two traffic streams have very different profiles, different Quality of Service requirements (QoS) and different Quality of Experience (QoE) requirements. The purpose of this research work is to understand how MAC layer coordination mechanisms compare against each other in carrying Voice Over Internet Protocol traffic in the presence of FTP traffic. This work is relevant because as operators deploy more and more Wi-Fi systems and the migration towards Wi-Fi continues, the choice of MAC coordination schemes will affect the Quality of User experience, especially on real-time/interactive services. An understanding of how MAC layer protocols/coordination schemes perform can enable development of better algorithms to support real-time services.

1.6.2. Key Research Questions.

In light of the fact that VoIP services are moving away from circuit switched networks to packet switched networks and that Wi-Fi is assuming an important role as a small-cell solution to congested and costly cellular systems. Moreover, in VoIP systems a voice call will is no longer guaranteed data rate and dedicated end to end resources, but rather it will use channels that are shared with other traffic types. **Are the Wi-Fi Medium Access technologies in their present form capable of carrying-out this task? Which MAC technology works best and under what circumstances? Can Medium Access enhancements improve performance? Does it matter which codec is chosen to support VoIP services?**

1.6.3. Scope

Although there are more recent standards such as the IEEE 802.11n and IEEE 802.11ac, the study is limited to the IEEE 802.11g because this is still the standard with the most deployed devices. Furthermore, the freely available research tools do not support these more recent standards.

This investigation is also limited to performance of MAC layer schemes, therefore effects of the physical layer factors such as Signal to Noise (Ec/No) and Inter-Symbol Interference are outside the scope of this work and can be investigated under a different research topic.

1.6.4. Contribution of this work.

This work was motivated by the increasing role of Wi-Fi networks in the Telecommunication ecosystem that has been dominated by cellular networks. This work also recognises that with the current network migration towards a flat All IP data network architecture, Wi-Fi networks will be required to support Voice over IP and other data services for multimode devices over a multiservice Heterogeneous Networks. Furthermore this work recognises the need to understand the behaviour of the Medium Access Control layer of Wi-Fi networks because it is key to satisfactory performance of Voice over IP over Wi-Fi when there are other competing traffic types.

Therefore the contribution of this work is as follows:

- An insight into the performance of PCF, EDCF and vanilla DCF under different VoIP and FTP traffic loads i.e 50/50 and 75/25 over an IEEE 802.11g network.
- An insight into PCF performance when only VoIP stations participate in the Contention Free Period.
- An insight into PCF performance when only VoIP Stations participate in the Contention Free Period, but with a lengthened Contention Free Period.
- An insight into performance of DCF, EDCF and PCF when fragmentation is used.
- An insight into increasing the buffer length to be the same length as the fragment.

1.6.5. Constrains

Although it was the researcher's desire to investigate the research questions with the latest IEEE 802.11 standards, namely IEEE 802.11n and IEEE 802.11ac, the freely available simulation tool (OPNET) does not support these standards. However, the researcher is of the view that, because the Medium Access Control MAC) layer has not changed much from one standard to the next, using the IEEE 802.11g should still give sufficient insight into the performance of this layer.

The investigations are only carried out in simulation. Although a testbed implementation would give mode realistic insights, it is currently not possible this is because of the following reasons:

- Only Atheros AR500x range of drivers provided open source drivers. These drivers allowed the research community to develop Mad-WiFi drivers which enable manipulation of the MAC layer parameters [47].
- The Mad-Wifi project was abandoned as a result only older generation of Atheros chipsets support these driver and the devices are difficult to source [47].

- The testbed implementation would have required at least 60 Wi-Fi capable devices (laptops, smartphones or a mixture). These devices are not readily available in the University's research lab.
- No manufacturer has implemented PCF because it is an optional feature, therefore a testbed implementation would exclude PCF from the investigations, and this is not desirable.
- OPNET is used extensively in the telecommunications industry by network operators, research institutions, academic institutions and device manufacturers for research and evaluation purposes. Furthermore, OPNET derived results are routinely accepted by prestigious and reputable institutions such as the Institute of Electrical and Electronics Engineers (IEEE) [48]. Therefore, it is the researcher's view that OPNET results are sufficient to provide an insight into the performance of these technologies. As such, the conclusions drawn from this simulation are a relevant contribution to the body of knowledge.
- The duration of this research does not allow sufficient time for a testbed implementation.

1.6.6. Organisation of the Report

The rest of this report consists of 6 chapters and they are organised as follows:

Chapter 2: This chapter gives a detailed account of Wi-Fi and delves into the working details of the Medium Access Layer technologies.

Chapter 3: This chapter will present a detailed account of VoIP and FTP.

Chapter 4: Once the technologies have been discussed, chapter-5 will outline the literature review and provide a background to some of the work that has been carried out in this area.

Chapter 5: This chapter will present the design of the simulations, relevant parameters and assumptions.

Chapter 6: This chapter will present the simulation results and key findings.

Chapter 7: This chapter concludes the work and recommends future work.

2. Chapter 2: Medium Access Control in IEEE 802.11 Wi-Fi/WLAN systems.

2.1. Background

In any telecommunications systems the channel/medium is shared by several devices because, it is a limited and expensive resource. Therefore, MAC mechanism need to designed such that they can afford all the users a fair chance to access the medium. Although the Wi-Fi spectrum is free it is also limited, thus these principles also apply to the Wi-Fi networks. The WLAN architecture was designed to be fault tolerant and to eliminate potential bottlenecks that a centralised approach would introduce. Although it uses a centralised Access Point as was introduced in 1.2.1, most of the medium access control was designed to be distributed so that the power to access the medium lies with the mobile station/Wi-Fi client [8]. This section of the report presents technical details of these mechanisms.

In the Wi-Fi network a bandwidth of 22 MHz for IEEE 802.11b/g and 20MHz for IEEE 802.11n. In order to ensure optimal usage of the channel resources, layer-2 from the OSI reference model also known as the Medium Access Control (MAC) layer manages how each user gains access to the medium. The MAC layer refers to a collection of mechanisms that control devices/users transmission of the air medium while minimising the collision between data from different users [16].

The MAC layer of the WLAN achieves this function in two ways, the distributed techniques and centrally coordinated techniques [8]. The distributed techniques put the power to access the medium in the device, thus the device itself has to contend with its neighbours for equal access to the medium; this scheme is called the Distributed Coordinated Function (DCF) and its enhancement called the Hybrid Coordination Function (HCF) [8]. In the centrally coordinated techniques the designated station, which is usually the Access Point (AP), controls access to the medium; this is called the Point Coordinated Function (PCF) and its enhancement that is referred to as HCF Controlled Coordination Access (HCCA). Both of these coordination mechanisms provide access to the medium while avoiding collision between data of different devices. The DCF forms the basis for any access control mechanism in Wi-Fi networks, and it is referred to as Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), see Figure 15. Other mechanism such as the PCF, Hybrid Coordination Function (HCF), HCF Controlled Channel Access (HCCA) also referred to as Enhanced Distributed Channel Access(EDCA) are implemented as an addition on top of the DCF(see Figure 15) [8]. This section presents the details of the various coordination methods for Medium Access in IEEE WLANs.

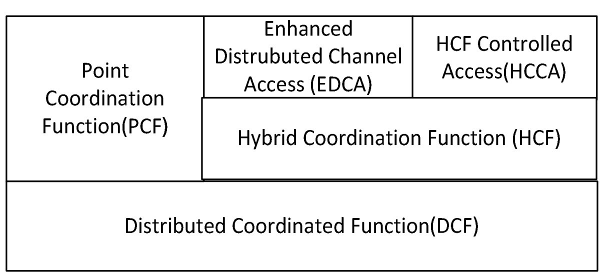


Figure 15: Wi-Fi MAC Architecture from [4]

2.2. Carrier Sensing

Wi-Fi standards are in a way derivatives of the IEEE 802.3(Ethernet Standards). The Ethernet technology uses Carriers Sensing to detect if there are any ongoing transmissions in the channel. The Wi-Fi network adopted this technique albeit with slight modifications such as collision avoidance, to a wireless environment where collision detection is not really possible [8]. Carrier sensing in Wi-Fi networks can take two forms, namely the physical carrier sensing and the virtual carrier sensing. The Physical carrier sensing is one whereby the device eavesdrops the channel for any ongoing transmissions, this eavesdropping is carried out by the physical layer (PHY).

The Virtual carrier sensing is carried out by the MAC layer. This carrier sensing is called the Network Allocation Vector(NAV) [8]. The NAV is a timer that is used for virtual carrier sensing, by telling other stations that the channel is busy for the specified duration even when the station cannot detect any carriers. This is necessary in the case where the sensing station is far from the transmitting station and the phenomenon is referred to as the hidden station phenomenon (see section 2.5 for a discussion of the hidden station phenomenon). The NAV is set by the duration value in each MAC frame header and indicates that the medium is booked for a certain period [4]. Carrier sensing employs both methods and if either method declares the medium busy, then the medium is declared busy and relevant mechanism are employed to manage data transmission.

2.3. Random Back-off

The Wi-Fi medium access techniques employ what is known as a Random Back-off procedure for Collision Avoidance (CA) in CSMA/CA. The Random Back-off procedure works as follows [4]:

- When the station detects that the medium idle (using the techniques in 2.2, for a duration that depends on the coordination scheme it enters a Back-off period to avoid collision with other stations that may be listening to the medium and wishing to transmit data.
- The Back-off period starts with selecting the minimum Contention Window (CW_{min}) value.
- The CW is sub-divided into time slots as shown in Figure 16, where the Slot-duration is dependent on the underlying physical layer standard and it takes the values 20µs for IEEE 802.11b and 9µs for IEEE 802.11a/g.
- The station then selects a random() number that falls in the range, 0 < random() < CW, where $CW_{min} < CW < CW_{max}$.
- The back off period is calculated using the random () value as per Equation 1 below.

Back-off timer = random() x Slot_Duration (Equation 1) [4]

- Each time the station is unable to transmit then it will jump to the next CW value (see Figure 16), where the next CW is calculated by the formula below values [8].

$CW = [0,1,3,7 \ldots 2^n-1]$ (Equation 2) [4]

- CW: the Contention Window

- n: A whole number and its maximum value depends on the standard

- For each failed transmission, the CW will increment until it reaches maximum, which is an indication that the AP is congested.
- The CW remains at maximum to allow the network to stabilise under congestion.
- After a successful transmission, the CW is reset to 0 and the process is repeated for the next frame.

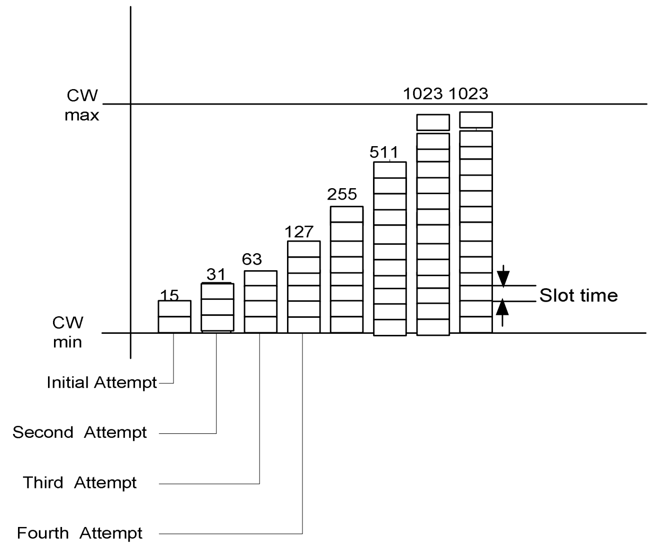


Figure 16: Incrementing Contention Window until it reaches maximum. [4]

2.4. Distributed Coordination Function (DCF)

DCF is fair coordination function that gives all the stations an equal chance of transmitting their data. The equality is achieved through a fair contention mechanism. DCF works in infrastructure mode. In this mode, all the transmission is relayed via the AP.

DCF operates as follows:

- When a device such as Stations 2, Station 3 or Access Point in Figure 17, has data to transmit, it listens to the transmission medium, using the techniques that were described in section 2.2, for any on-going transmission.
- If the medium is busy the station will postpone its transmission until the medium is idle for duration of at least DCF Inter Frame Spacing (DIFS). The value of DIFS is determined by the underlying layer 1 standard.
- If the station detects that there is no on-going transmissions for at least DIFS, the station will pick the lowest contention window and start the Random Back-off, as described in 2.3. This Back-off process is started in order to avoid collision see Figure 17 below for four stations with data ready to transfer.
- When the station is in the Back-off state, it continues to check if the medium is busy/idle for each slot duration. The slot duration depends on the standard and it is 20ms for 802.11b and 9ms in 802.11a/g [8].
- If the medium is idle for the duration of the slot time, the Back-off timer is decremented by a slot duration, i.e. 9ms for IEEE 802.11g.
- If the medium is busy the Back-off is not decremented.

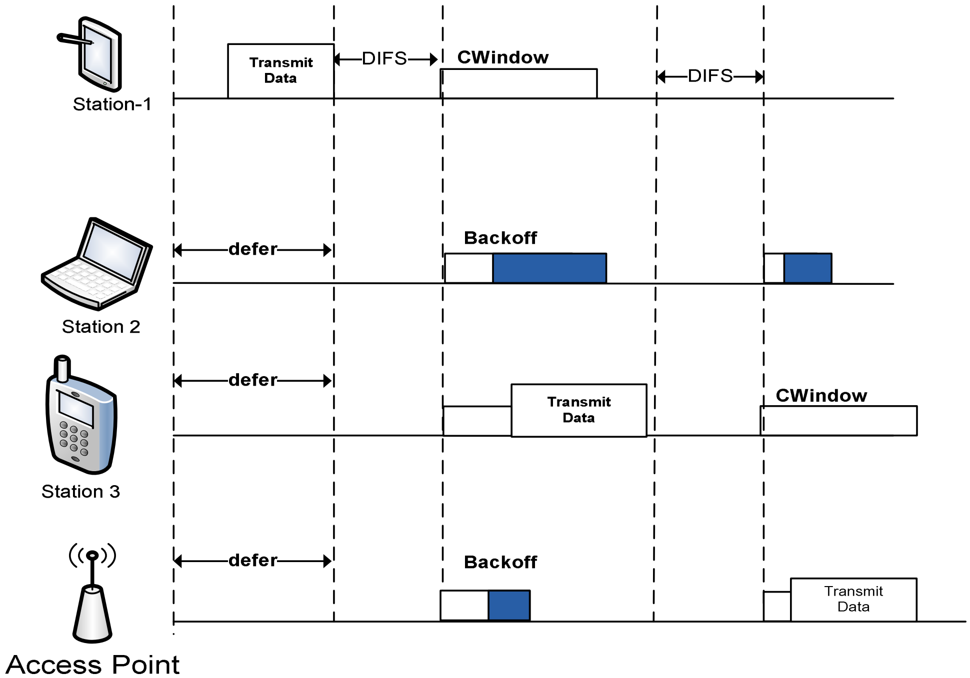


Figure 17: DCF Transmission when the channel is idle [4]

When the data transmission is completed successfully as indicated by the correct Frame Check Sequence checksum, the AP will wait a short period called Short Inter Frame Spacing (SIFS), then transmit a 14-byte Acknowledge (ACK) frame to confirm. However, if the Frame Check Sequence is incorrect, the AP will not send a No Acknowledge frame (NAK). The IEEE 802.11b/g/n (2.4GHz) standards specify values of 10µs for the SIFS, while the IEEE 802.11a/n(5GHz)/ac specify 16µs.

In the unlikely case that two stations start to transmit at the same time as in Figure 18 below, a collision will occur. When a collision has occurred, both stations will not receive an ACK frame SIFS seconds after the transmission was completed i.e. the ACK waiting timer times out. Both stations will enter a back-off state as in 2.3 and pick the next highest CW and repeat the Back-off process.

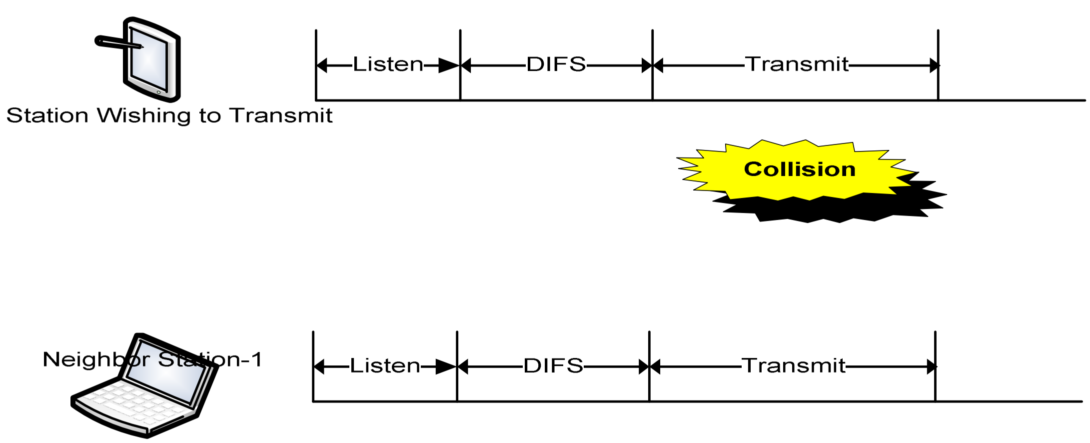


Figure 18: Simultaneous transmission

The station that randomly picked the shortest back-off timer will transmit first. The likelihood that two stations can pick the same back-off value is diminished by the randomness of the contention process. This procedure minimises collisions even under high load condition [8]. Although the method promotes fairness amongst stations, it is unsuitable for supporting real-time services such as VoIP, because the real-time services need to be prioritised over other services; otherwise the user experience becomes poor and the service becomes unattractive to the end user [8]. As thus, DCF in its conventional form may not be suitable for service VoIP and FTP station and this investigation seeks to answer the question, under what circumstances is DCF suitable or not suitable for a mixed station BSS.

2.5. Hidden Station Phenomenon

The notion of virtual carrier sensing by means of Network Allocation Vector (NAV) was introduced earlier in section 2.2, this section expands this notion further. The NAV is used to alert station that the medium is currently busy [4]. It is sent in the case of a Request to Send/Clear To Send (RTS/CTS) mechanism, Figure 19. In the outlined scenario:

- Station-2 listens for any transmission on the medium and determines that the medium is idle for DIFS period.
- Station-2 begins its transmission to the AP.
- At this time Station-2 is unaware that Station-1 is in the middle of a transmission because they too far apart from each other that the transmission from Station-1 cannot be heard by Station-2.
- This leads to a collision.

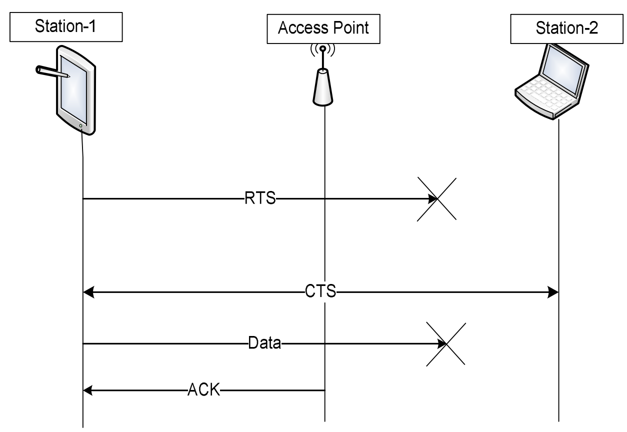


Figure 19: RTS/CTS mechanism. **[6]**

This is called the hidden station phenomenon, wherein all stations can hear the AP, but two stations are too far apart to hear each other. Therefore, if Station-1 wishes to transmit it can request the medium by sending a Ready To Send (RTS) frame to the access point with a Network Allocation Vector, indicating how long the station needs the transmission resources for [8]. The Access Point will then send a transmit jam signal, Clear To Send (CTS) to all stations thus allowing the requesting Station-1 to transmit and barring all other stations including the AP from transmitting. This value of the NAV is the same value that is used during the Virtual Carrier sensing as mentioned above. The RTS/CTS mechanism complements the CSMA/CA and ensures that under all circumstances the CSMA/CA medium access mechanisms holds. If CSMA/CA falls apart, as it would in the case of hidden stations then the DCF, PCF and EDCF mechanism would also not hold.

2.6. Point Coordination Function (PCF)

PCF is an optional coordination scheme for IEEE 802.11 standards. It is a centralized coordination access mechanism, wherein stations take turns to transmit depending on their position in the polling list [8]. This technique ensures that only one station can transmit at any time. This means there is no contention for access to the medium but rather the stations are scheduled by the AP for transmission, hence it is called Contention Free access method [8].This technique was the first attempt to enhance Wi-Fi to support real-time services. Although an interesting prospect, PCF was never widely adopted by WLAN manufacturers, hence its performance has not been widely investigated [8]. PCF is an optional function during the DCF operation. When PCF is in operation, the period is referred to as the Contention Free Period (CFP), during this period, DCF contention based mechanisms are temporarily suspended and stations are allowed access to the medium according to the polling list [8].

During the contention free period, the RTS/CTS mechanism is not used, but the Point Coordinator (PC) polls all the pollable stations. The PC is the station that allocates other stations in a BSS a transmission slot in a round-Robin fashion, usually the Access Point. When polled, the station will transmit only one frame. If the transmitted frame is not acknowledged by the PC, the Station will not retransmit unless it is polled yet again. If the polled station is not PCF capable, it will indicate to the PC via DCF mechanism that it is not pollable and PC retains control of the access medium [4]. Furthermore, the PC can use the Contention Free Period to transmit frames to stations and not necessarily to poll them. As a result, PCF stations have a higher medium access priority than DCF stations, when both types are operating within the same BSS.

To begin a Contention Free Period [4]:

- The Point Coordinator (PC) carrier senses the medium according to the procedure that was described in 2.2.
- If the PC declares that the medium idle for a period equalling the PCF-Inter Frame Space (PIFS), the PC sends out a beacon frame containing the Contention Free (CF) parameters and DTIM, Figure 20.
- When a DCF based station receives a DTIM frame, it responds with a DCF ACK and the PC retains control of the transmission medium.
- The PC then waits a further Short Inter Frame Spacing (SIFS) period; shortly thereafter a frame is transmitted.
- The transmitted frame can be a Data Frame, CF Polling Frame; Data with CF Poll Frame, Management Frame, and Contention Free end Frame. This data is destined to a station that is participating in the CFP.

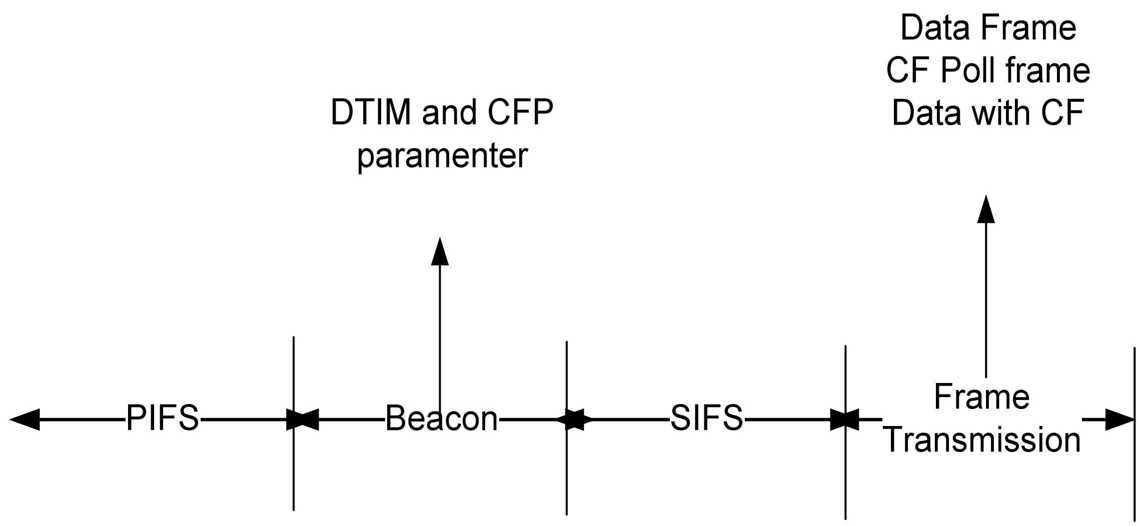


Figure 20: Contention free Period Timing [4]

The Contention Free Period (CFP) alternates with the Contention Period (CP). During the Contention Period, DCF is in control of the medium access. The Contention Free Period begins when a beacon frame Delivery Traffic Indication Map (DTIM) is sent. This DTIM is transmitted periodically, see Figure 21 below. The duration and period of the Contention Free Period are determined by the PC. Although, the duration of the Contention Free Period varies with the maximum set via a user determined parameter, the period of the CPF is not constant. Therefore, the PC can terminate the Contention Free Period before the maximum duration is reached depending on the size of the polling list and the traffic requirements [4].

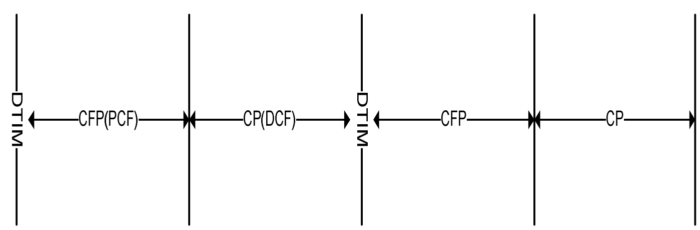


Figure 21: DTIM transmission **[4]**

The DTIM beacon is sent to all stations at the beginning of the period. This includes stations that do not support PCF. On receiving this beacon frame the stations set their Network Allocation Vector (NAV) to the value equal to Contention Free Period (CFP).

The PC can be configured to support Contention Free Period for sending and receiving frames. When the PC is operating in transceiver mode during CPF, it needs to maintain a polling list. The station needs to get itself enlisted as CF-pollable during association. If there are entries in the polling list, the PC will poll a subset of stations according to the Association Identifier(AI) in ascending order [4], where the AI is a control and management identifier that the station is assigned when it attaches to an Access Point. In order to optimise usage of the channel, the polling frames are normally packaged with data and acknowledgement frames.

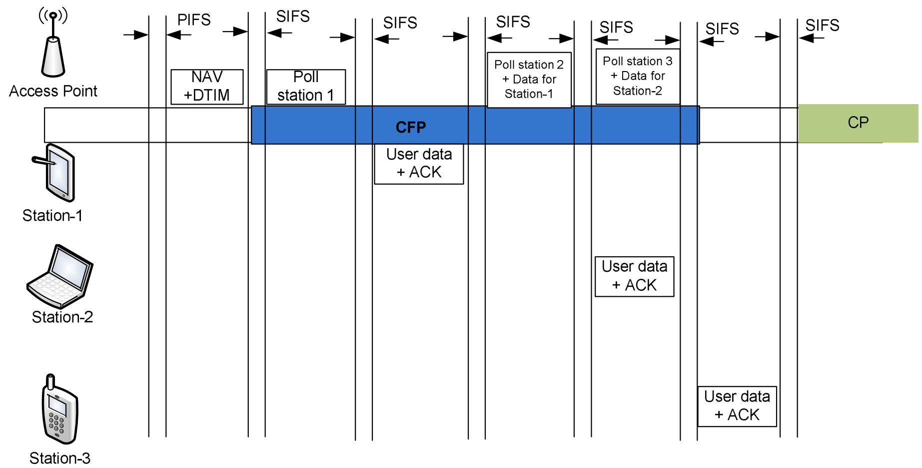


Figure 22: PCF management example. [4]

The diagram in Figure 22 summarises the polling mechanism of PCF. This mechanism can be summarised as follows [4]:

- The Access Point (AP) is the Point Coordinator (PC).
- The AP/PC senses the medium for any transmission and declares that the medium is idle for PIFS.
- The AP/PC sends a jam signal (DTIM) to all the stations indicating how long the medium will be jammed for (NAV), i.e. the duration of the Contention Free Period (CFP).
- The AP/PC sends the first polling frame indicating that Station-1 is clear to send data.
- Station-1 sends its data plus a frame acknowledging the poll (ACK).
- A short period SIFS is allowed to pass and the AP/PC sends its data that is destined for Station-1 along with a polling frame that gives Station-2 permission to send its data.
- Station-2 sends its data along with an ACK.

In this study we investigate the suitability of this PCF mechanism for carrying VoIP data when there is FTP data in the same network. The study investigates the effect of allowing all stations both VoIP and FTP to participate in the CFP. Then only VoIP stations in the CFP while lengthening the CFP to allow VoIP stations more time to transmit.

2.7. QoS Enhancement in 802.11

The WLAN technology was not initially designed to support real-time services such as voice, video streaming and gaming. PCF was the first attempt to support real-time services over WLAN. Since PCF several attempts have been introduced to enable WLANs to support services with stringent QoS requirements.

The QoS enhancement standard IEEE 802.11e retained both the Contention Based (CB) and the Contention Free (CF) paradigms in a new technique called the Hybrid Coordination Function (HCF) [8]. In addition to CF and CB, it introduced techniques to ensure that stations that carry services requiring stringent QoS get a higher probability of winning the contest for the medium and retaining access to the medium for longer [8]. The standard introduced differentiation of services in order of Access Category (AC), where Voice (AC-1), Video (AC-2), best effort data (AC-3) equivalent to DCF and background data (AC-4). The services with top Access Category such as voice are prioritised and background traffic receives the best effort service with the least priority.

The polled (PCF) and contention (DCF) based mechanisms were further enhanced with other techniques including streaming/burst mode [8]. In this technique the station is allowed to maintain connection and transmit several frames. The other methods are the block acknowledgement, where instead of individual acknowledgement of frames, one ACK frame is used to acknowledge several successfully transmitted frames, thus reducing the overhead that results from acknowledgement for every frame. The other enhancement technique in HCF was to send no acknowledgement frames at all, this works in applications that have a high tolerance for data-loss but low tolerance for latency [8]. The next few sections will cover the various QoS enhancements in WLANs.

2.8. Hybrid Coordinated Function(HCF)

2.8.1. Background

The HCF technique introduces two QoS mechanisms namely, the Enhanced Distributed Channel Access (EDCA) and the HCF Controlled Channel Access (HCCA) [8]. The EDCA is an enhancement of the DCF method, but it allows a selected station to win the contest for the medium unfairly. HCCA is an enhancement of the PCF method where allocation of the transmission medium is centralized to the AP and access is granted on a polling basis. The key enhancements to these methods are the introduction of Transmission Opportunity (TXOP), Arbitrated Inter Frame Spacing (AIFS) and changes to the selection of back-off parameters [8]. These adaptations are different for each access category. In order to ensure that both the transmitter and the receiver devices are aware of the QoS requirements all the frames from stations that support QoS will include an additional QoS field. Although the QoS mechanisms ensure priority treatment for real-time services, stations with real-time requirements have an equal chance of gaining access to the medium.

2.8.2. Enhanced Distributed Channel Access(EDCA)

Unlike in the classical DCF, EDCA does not treat all user data equally. It uses a discrimination method wherein the type of user data is given an Access Classes (AC). Table 7 summarises the different AC that are defined by the IEEE 802.11 standards. The lower value of 1 for User Priority (Access Class Background (AC_BK)) indicates that the service does not require high priority treatment, whereas the higher value of 7 for User Priority (Access Class Voice (AC_VO)) indicates that the voice service needs to be treated with urgency.

Table 7: Access Class priorities in 802.11 from **[4].**

	User Priority	802.1D designation	Access Class	Designation
Lowest	1	BK	AC_BK	Background
	2	—	AC_BK	Background
	0	BE	AC_BE	Best Effort
	3	EE	AC_BE	Best Effort
	4	CL	AC_VI	Video
	5	VI	AC_VI	Video
	6	VO	AC_VO	Voice
Highest	7	NC	AC_VO	Voice

In the classical DCF method, section 2.4, every station in the BSS had to wait DIFS time for the medium to idle before going into Random back-off and subsequently attempting to transmit data. The stations with a higher value of User Priority have a shorter DIFS value; therefore they will always gain access to the medium ahead of station with a smaller User Priority value.

DCF manages collisions by giving all the stations an equal chance of picking a shorter back-off interval. The EDCA enhancement ensures that the stations pick their back-off intervals in order of the Access Class of frames in their queue, with User Priority 7 stations picking the shortest back-off intervals and User Priority 0 picking the longest intervals. This selection mechanism ensures that stations with higher User Priority will always be the first to get serviced.

TXOP is defined as maximum time that a station can occupy the transmission channel while transmitting a series of frames and their corresponding acknowledgements [8]. TXOP is be allocated to a station in the contention based mechanism (EDCA), when the station wins the contest or by the central coordination method(HCCA) where the station receives a Quality of Service poll frame [4]. It is measured in 32µs increments. In the EDCA method, the stations with a higher User Priority frames to transmit will receive a longer TXOP that the stations with lower User Priority, thus occupying the medium for longer and transmitting more frames to meet the user's requirements. Therefore TXOP is defined on per Access Class/User Priority Basis.

There are two EDCA Transmission Opportunity (TXOP) modes namely the initiation TXOP and the multi-frame transmission within an EDCA TXOP [4]. The initiation of TXOP refers to the TXOP when the station is first granted access to the transmission medium for exchanging a single frame. However, the station can be granted permission to transmit additional frames; this is referred to as the multi-frame TXOP. The multi-frame transmission is only permitted if the time required to transmit the next frame does not exceed the remaining duration of TXOP. Furthermore, the frame in the queue has to be of the same User Priority/Access Class as the Access Class for which the TXOP was assigned. The duration of the TXOP is specified by the assigned TXOP limit.

When the TXOP holder has been granted a TXOP limit of '0' it can do the following [4]:

- Transmit a single data frame.
- Transmit Acknowledgements.
- Transmit an RTS/CTS frame.
- Transmit a link adaptation frame.

In the event that the TXOP limit is a value other than zero, the transmitting station needs to breakdown the frame such that amount of time required to transmit this frame, at the currently selected data rate, does not exceed the allocated TXOP. An example, if the station has a frame of 1024 Bytes/1 MB in the queue and based on the radio conditions the selected transmission rate is 1 Mbps, then the time it will take to transmit this frame is 1s=MB/1Mbps. If the station has been assigned a TXOP limit of 10ms, then the maximum frame size that can be transmitted is 10.24Bytes = 10ms x 1Mbps. Therefore, the frame must be fragmented (broken-down) to at most 10.24 bytes to avoid exceeding the TXOP.

The Back-off mechanism for EDCF is similar to the generic Back-off timer as described in section 2.3. There are however some notable differences:

- The CW is assigned on the Access Class basis.
- A new interval called the Arbitration Inter Frame Spacing (AIFS) is defined per Access Class.

- The services with a higher User priority such as voice have a relatively shorter AIFS compared with other services.

The figure below Figure 23 highlights the difference between AIFS and Inter Frame Spacing for other coordination schemes.

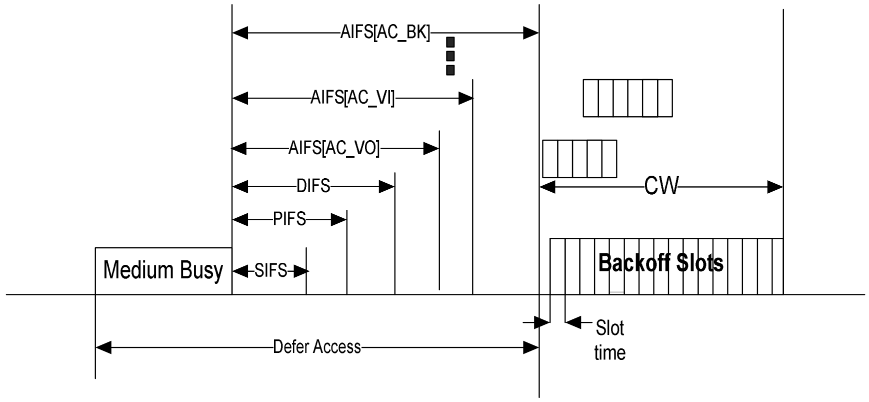


Figure 23: Relationship between the Inter Frame Spacing Intervals from **[4]**

The above parameters assume the following values for the IEEE 802.11g:

- Slot = 9µs
- SIFS =16µs
- PIFS = 25µs
- DIFS = 34µs
- AIFS >=34µs (This value is small for high User Priority services and large for low User Priority).

2.8.3. HCF Controlled Channel Access(HCCA)

HCCA enhances the PCF for support of applications/services with stringent QoS requirements. In this mechanism, the Point Coordinator (PC) is referred to as the Hybrid Coordinator (HC). In HCCA, the HC is given more powers over access control, similar to PC as was done with PCF. The HC - which can be any station although it is usually the AP- can allocate the TXOP to stations in accordance with the AC of the traffic in their queue. This TXOP is sent to the station during polling and it is protected by the NAV which acts as a virtual carrier during the carrier sensing phase by other stations. The stations can request TXOP during the polling phase. The HC also has a higher priority to access the medium during the contention period. The CFP is created by means of a NAV as was the case with the PCF. If the station no longer requires the medium, it can notify the HC and be removed from the polling list. HCCF performance is not investigated in this work because it is not available in the simulation package.

2.9. Block Acknowledgement and No Acknowledgement

The block acknowledgement mode attempts to optimize channel utilization by group acknowledgement of several frames, instead of sending an Acknowledgement for each frame, see the Figure 24. The block acknowledgement works in two modes, the immediate mode and the delayed mode. In the immediate mode, the teardown sequence is completed immediately, whereas in the delayed mode, the last frame exchange of the teardown from the receiver station can be delayed. IEEE-802.11 [4] recommends immediate block acknowledgement in high bandwidth and low latency service requirements. Delayed block is recommended for operation in non-latency sensitive applications.

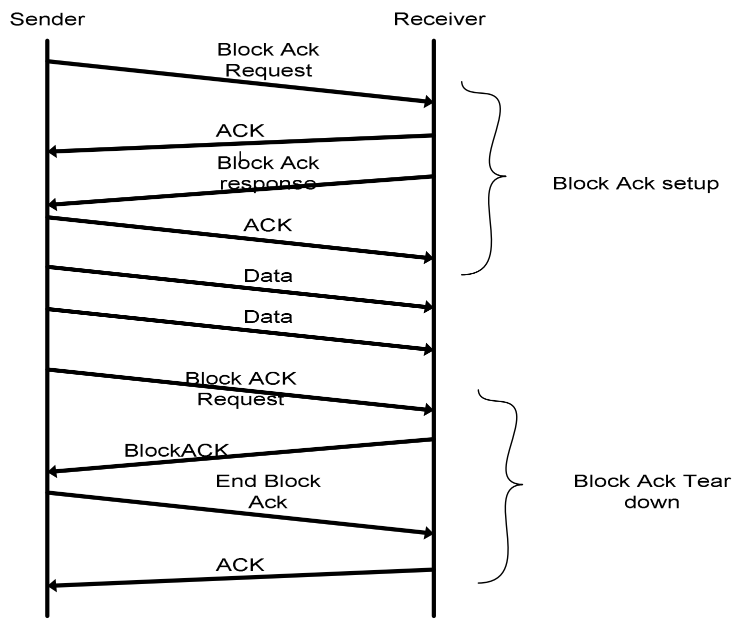


Figure 24: Block acknowledgement. **[6]**

Before block acknowledgement can be used, both stations need to ensure that they are capable of block acknowledgement. Therefore the sending station will send a request to indicate its desire for block acknowledgement and the receiver will indicate its ability to perform block acknowledgement. This mechanism works as an enhancement to the techniques discussed above, namely PCF, HCCA, RTS/CTS and HCF.

The no Acknowledge policy may be used to reduce the latency. However, due to the fact that it may potentially introduce frame loss, it works as a complementary mechanism to RTS/CTS, PCF, HCCA and EDCA. It is a suitable option for these mechanisms because they ensure that only one station is transmitting during the allocated period, hence it is less likely that a frame will experience collisions. Moreover, resending of lost frames in the voice network is almost illogical as voice frames need to be processed in sequence. In this study we also investigate the suitability of this mechanism for supporting VoIP services in the presence of competing data traffic.

2.10. Fragmentation

Fragmentation of frames means breaking down a large frame into smaller sub-frames before transmission over the medium. It is commonly used to manage data transmission over the medium. It is especially usefully in unforgivingly lossy wireless media which suffer from Inter Symbol Interference (ISI), fading, co-channel interference etc. Fragmentation seeks to manage these shortcomings from the medium access layer without making any assumptions about the quality of physical layer. Therefore, it can reduce the losses as well as reduce the impact of a lost frame. However, fragmentation comes at an added cost because each fragment (piece) of the frame needs to be padded with headers that contain fields such as addresses, control field, sequencing and error checking. This overhead is 34 octets in IEEE 802.11 MAC frame [17]. This padding introduces overheads on the network and reduces efficiency. This reduction in efficiency can introduce additional delays, but at the same time it can reduce packet-loss. Therefore fragmentation is a balancing act between the size of the fragment and quality of the underlying physical medium. The IEEE 802.11 medium access dictates that all the fragments must be of equal length except the last fragment which may be shorter [4].

We extend our investigation of the Medium Access Control schemes above by investigating two fragment sizes of 256KB and 1024 KB for each of DCF, EDCF and PCF under the various loading conditions.

3. Chapter 3: VoIP and FTP Traffic

3.1. VoIP

3.1.1. Introduction

The voice service was for the longest time provided by circuit switched networks, notably Public Switched Telecommunications Networks (PSTN), [49]. This system was tailored specifically for providing voice services. As a result before communication can begin transmission resources are reserved along the call route between the initiating party and the recipient, by the intelligent core network. These resources are retained for the duration of the call and are at no time available to other users regardless of whether the existing users are in fact actively utilising them [6]. This was an obvious inefficient usage of the expensive and scarce transmission resources, be it the long-haul communication trunks or the radio spectrum. Moreover, it was difficult to enrich the voice experience with other services as the architecture was tailored specifically for voice without much regard for other services that could enhance the communication experience such as presence, location, availability, device capabilities, video calling, instant messaging [50].

Data networks or the packet switched domain developed in parallel to circuit switched networks. In the data paradigm, emphasis was placed more on intelligent end-systems and a "dumb" packet forwarding core network. Advances in the data packet networks and especially TCP/IP architecture brought with it the possibility to provide all services on a single flat architecture that is independent of the lower layer bearer, see Figure 25 below.

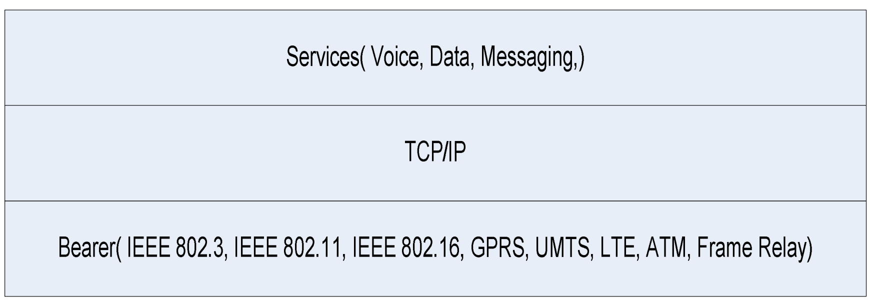


Figure 25: High level VoIP Protocol Stack

This meant that it was no longer necessary to tailor the lower layer bearers for a specific service, but rather to support IP. Therefore, it was only natural that the voice service was migrated as one of the services over the IP architecture, which lead to the development of Voice over IP (VoIP). This would inherently result in cost reductions as the telecoms operators and enterprises would only need to deploy and support a single multi service network. However, data networks were also not traditionally tailored to provide real-time services such as voice, but rather best effort services. Real-time services have stringent quality of service requirements because of their interactive nature. Therefore, to support real-time services such as VoIP further QoS enhancements services were required. The next few sections will cover the details of VoIP service, architectural overview, codecs, VoIP implementation, quality of service and finally the tools for objective and subjective evaluation of voice.

3.1.2. VoIP Protocol Stack

Before delving into the operation of a VoIP system, it is imperative to understand the protocols that make VoIP possible. This section will describe some of the key protocols that are involved in the VoIP system. The actual operation of the protocols will be expanded in the subsequent sections. Discussion of the VoIP architecture in this section assumes that the reader is familiar with the concept of Open System Interconnect (OSI) layered abstraction as well as TCP/IP protocol stack.

VoIP protocols fall into three categories, the data plane protocols, the control plane protocols and the gateway control protocols [51]. The function of the control plane protocols is establishment, maintenance and tearing down of a VoIP call. The function of the data plane protocols is to carry the voice streams between the speaker and listener. This set of protocols will be discussed separately.

The gateway protocols ensure interconnection between IP based networks and legacy networks. The Figure 26 below shows the generic protocol stack for VoIP. VoIP protocols both control plane and data plane are carried over User Datagram Protocol (UDP). Unlike TCP, UDP does not guarantee safe arrival of packets to the recipient [6]. This is not a concern because voice can tolerate some levels of data loss. Importantly though, transmission guaranteeing methods in TCP, such as retransmission of packets and flow control are unsuitable for voice applications. They introduce additional overheads, delay and variation in the arrival rate of packets at the sink. Furthermore, it wouldn't make sense to retransmit a lost voice packet as it would add no coherence to the conversation.

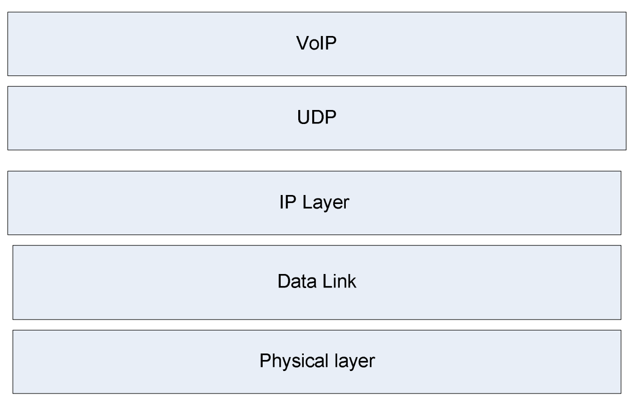


Figure 26: VoIP generic protocol Stack **[17]**

Once the VoIP packets have been encapsulated into a UDP datagram, they are carried over the IP network like any other packets, independent of the transmission medium. These packets are handed over to the network layer and are never acknowledged.

The Figure 27 below is adopted from [51]and it shows the details of the different VoIP protocol categories. The main function of the Real-Time Protocol is to identify the voice codec type and append sequence numbers and timestamps to allow correct processing of the voice packets at the receiving end [8]. The RTP sender appends a time stamp and a sequence number to the header of each VoIP packet. There is also a payload type that field that identifies the codec scheme used in the payload. The sequence numbers are necessary to ensure that during playback, the buffered packets are played in the correct order so that the speech is coherent.

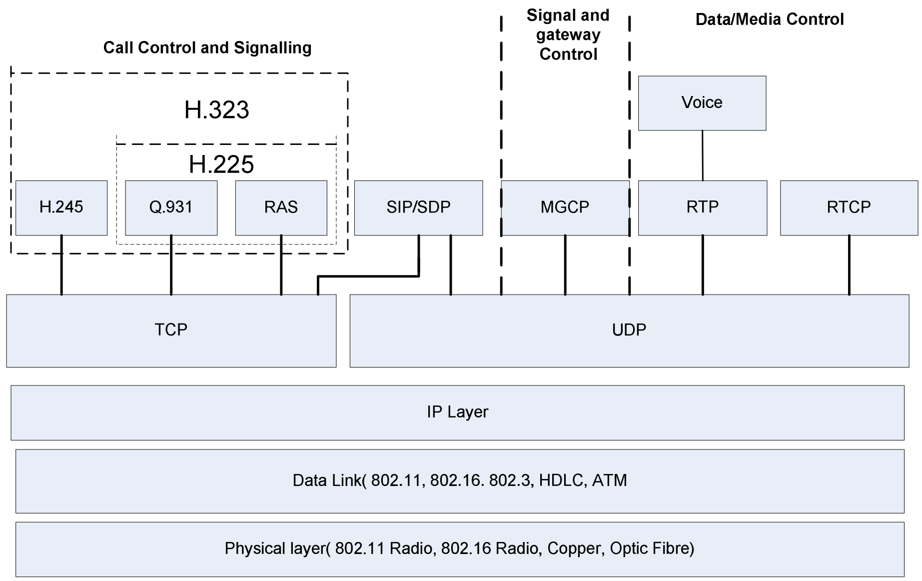


Figure 27: VoIP detailed protocol stack **[51] [2]**

The function of the Real Time Control Protocol (RTCP) is to exchange VoIP session performance data between the sending and the receiving stations; this report is exchanged every 5 seconds [8]. Amongst the data that is exchanged between the sender and the receiver of an RTCP is the packet loss rate, packet discard rate, burst length, end system delay, signal level, noise level, round trip delay, jitter buffer etc. Currently, IETF is working on RFC 3611 which will extend the list of reports shared between the stations.

Session Initiation Protocol (SIP) is a session establishment, maintenance and tear down protocol. It was developed by the IETF under RFC 3261-3264 [8]. SIP supports the following capabilities for establishment and termination of a connection:

- User location: The IP address where the user is located.
- User availability/presence:
- User End system capability: Determination of the media options available for connection.
- Session setup: Establishment of session parameters using the Session Description Protocol eg. Encoding scheme to be used, packetisation interval etc.
- Session management: Features for call transfer and termination.

SIP identifies the User Agents, which are the end system and the SIP proxy server. The SIP proxy server houses the functions of mobility management, signalling message relay during a call. The Session Description protocol (SDP) is defined in IETF RFC 4566 [52]. It is a protocol framework for representing the details of the media during a multimedia session such as VoIP, video streaming and multimedia teleconferences [52]. It enables sharing of session information such as Session name and purpose, duration of the session, the nature of media that will be shared during the session, addressing and formatting for exchange of the media, the rate at which the media will be exchanged [52].

The Figure 28 below outlines the SIP call flow for a two party session. As per diagram, the initiating party sends an INVITE message with a URI (address of the recipient) and a Session Description. The recipient then responds with a RINGING message code 180. Once the call has been accepted, the recipient returns a 200-OK message to indicate to the caller that the call has been connected. The caller then acknowledges the session and data exchange between the two parties can begin.

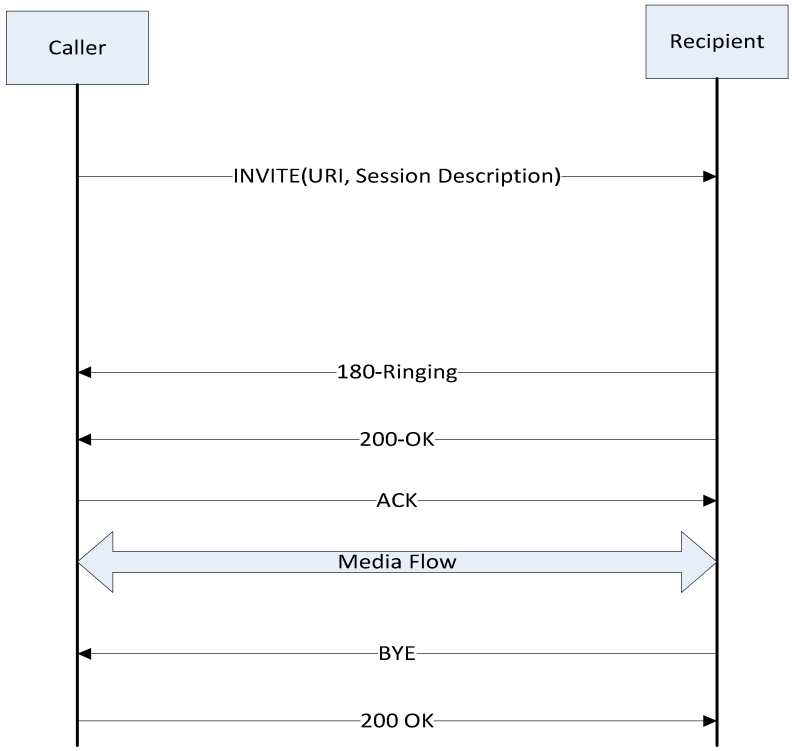


Figure 28: SIP Call flow excluding the proxy server and the location register. **[50]**

When one of the parties decides to end the exchange, they will send a BYE message to the other party and the other party will acknowledge a disconnection with a 200-OK message.

H.323 is another real-time control protocol with the same functions as SIP. The end systems use the H.245 protocol to exchange capabilities and preferred parameters such as the codec scheme. The H.225 describes a list of messages that establish, maintain and tear down calls. Because H.323 relies on an intelligent core to manage the calls, Registration Admission Status (RAS) function is to manage connection between the handsets and the gatekeeper in the core [53].

The main difference between the H.323 and SIP is that SIP relies on intelligent end systems, whereas H.323 relies on an intelligent core network. H.323 is one of the earliest real-time protocols, hence it retained the old paradigm of intelligent core and “dumb” end systems.

Although there are several transport protocol in IP networks, there are two key protocols that are commonly used namely the Transport Control Protocol (TCP) [54] and the User Datagram Protocol (UDP) [55]. The key difference between the two protocols is that TCP is referred to as connection oriented and reliable protocol whereas UDP is referred to as the connectionless and unreliable protocol [6]. UDP is said to be connectionless because it involves no prior handshake before data exchange. UDP effectively appends source and destination port numbers and a checksum (Figure 29) to an application layer datagram. Therefore UDP is a very lightweight protocol, because its overhead is only 8-bytes compared to a TCP header of 20 bytes [6]. UDP is therefore an unreliable transport protocol i.e. it gives no guarantees that the packet will arrive at their destination. As a result of this light weight and the fact that voice traffic is loss tolerant UDP is the preferred transport layer protocol for voice services. Furthermore, TCP’s reliable connection mechanism has built in congestion control and this creates jitter problems for real-time traffic.

Lastly, because of its connection orientation TCP cannot be used in multicast systems such as telephone and video conferencing. TCP is discussed in section 3.2 with FTP traffic.

Source Port	Destination port
Length	Checksum

Figure 29: UDP Header (from **[17]**)

3.1.3. VoIP System Operation

Figure 30 below highlights key elements of a VoIP system that uses SIP as the control plane protocol. The top down operation of this system is described below.

In the diagram below, user A wishes to communicate with user B. User A dials user B from a VoIP client, such as Skype or a SIP handset. Before communication can occur between the two parties, a SIP session needs to be established. When user A dials user B, the SIP protocol is called into action.

The SIP protocol has built in presence mechanism to determine the availability of the called party. This is managed by means of the Presence Agent [56]. Once SIP realises that party B is available to talk. The SIP protocol sends an INVITE message from A to B with A as the SIP source address. The INVITE message also includes the SDP with details of the requested call. When party B's phone rings, SIP return a code 200 to the sending party to indicate that the recipient's handset is ringing. The receiver then answer his/her phone and SIP returns an OK message back to the sender, indicating to the sender that connection has been established. The caller then sends an Acknowledgement of the session establishment. This completes the handshake [51].

During call establishment, the two stations agree on the speech codecs to use for the duration of the communication and the packetisation interval. Packetisation interval refers to the maximum length of speech data that may be packetized. Once the session is open the users are now ready to communicate.

When user A speaks into his/her phone, the analogue voice is digitized (Analogue to Digital Conversion). The digitized voice is then encoded with the selected codec. The different codecs are discussed in the next section of this report. The encoded speech data is then packaged into datagrams based on the selected packetisation interval.

According to [2] packetisation interval needs to be chosen with care because it involves balancing of the payload efficiency (Payload efficiency is the payload size/total packet size) and the delay/time taken to fill the packet. The shorter the packetisation interval, the shorter the amount of time required to fill it and hence the shorter the delay. This comes at the expense of payload efficiency, in that the total payload as a percentage of the packet is small, thus leading to inefficient use of the bandwidth. Therefore, the idea is to pack the most data possible, while causing the least possible delay. An uncompressed RTP/IP header is 40bytes [2], therefore, a 30byte payload equals efficiency of 43% i.e. 30/(30+40). The different codecs, still to be discussed, generate voice packets at different rates. An example is the G.711 codec, which generates packets at 64Kbps, hence it will generate a 30byte packet in 3.75 micro seconds, whereas another codec with the rate of 8Kbps will take 40ms. The increasing the packetisation interval can increase the number of calls that can be carried, but comes at the expense of increased delay which degrades the call quality.

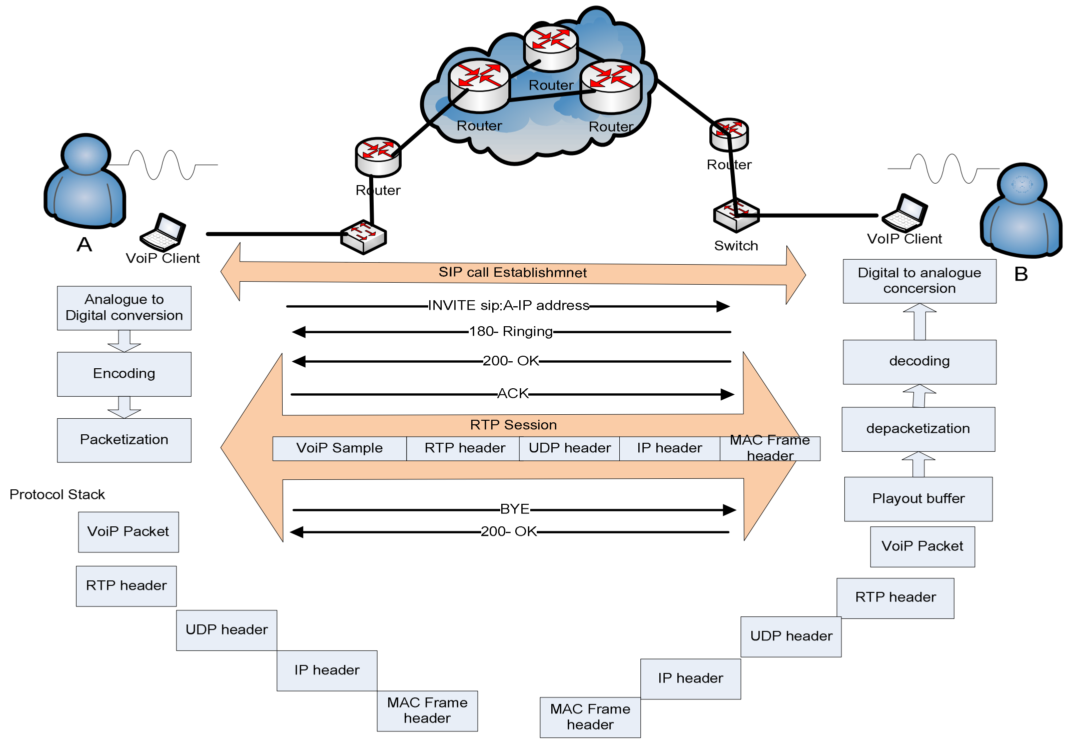


Figure 30: VoIP system operation [2].

The voice packet is appended with an RTP header (see Figure 31 below), the fields are described in [6]. This header contains amongst others, the timestamp, sequence number and the codec type. The RTP header is then appended with a UDP header. UDP header is short, 8-bytes; see Figure 29 and it provides only the delivery and checksum services, unlike TCP (20bytes) which offers additional services such as flow control, error control and sequenced delivery. This simplicity is necessary for streaming applications as it reduces the packet overheads and improves the streaming experience.

<table>
<tr><td>Version</td><td>Padding</td><td>Extension</td><td>CSRC Count</td></tr>
<tr><td>Marker</td><td colspan="3">Payload Type</td></tr>
<tr><td colspan="4">Sequence number</td></tr>
<tr><td colspan="4">Timestamp</td></tr>
<tr><td colspan="4">Synchronization Source Identifier (SSRC)</td></tr>
<tr><td colspan="4">Contributing Source Identifier (CSRC)</td></tr>
</table>

Figure 31: RTP header [51]

The UDP header is then appended followed by the IP header. The IP header helps the packet to traverse the network between the sender and the receiver, details of this header can be found in [50] [6] [17]. This header is the only one that is visible to the intermediate devices. The IP header is then appended with the MAC layer header. Furthermore, the packet now known as the frame with is appended with the PLCP header as was discussed earlier. The IP packet is then released into the network. UDP makes no provision for safe delivery of the packet, this is necessary in order to minimize the overheads.

At the receiver end the above process is reversed and the different headers are stripped and the packets are buffered for playback purposes. It was mentioned earlier that the UDP does not provide a sequenced delivery service, but this services is necessary for voice because if packets are played-back out of sequence then the conversation will become incoherent. As a result the sequenced delivery of packets service is provided by the VoIP protocol stack, specifically the RTP, see Figure 31.

The size of a playback buffer is key to the design of VoIP system. If the buffer is too short, all the packets received outside its size will be dropped, if the buffer is long it adds additional transit delay [8]. The main issue is that calls do not generate traffic at a constant and predictable rate for buffer, therefore fixed length buffers usually perform poorly. Dangerfield *et al* in [57] and Malone *et al* in [58] studied the effect of buffering on VoIP capacity. In [58] it was concluded that the size of the buffer should be proportional to the amount of VoIP traffic that the network is carrying. However, [59] concluded that there exists a minimum buffer size at which maximum calls are accommodated, below this value there is excessive packet loss. As there is no agreement over the optimal buffer size, at the moment the best practise is to use an adaptive buffer. The adaptive buffers adjust automatically based on the amount of jitter that is experienced during a call.

In some cases to increase efficiency of the payload, some header compression schemes are applied. Researchers in [60] studied the effect of introducing Robust header Compression (RoHC) for VoIP in IEEE 802.11, they reported a 23% improvement in voice quality when the network is heavily congested. However, they report no benefits as a result of RoHC when there is little or no congestion.

It should be noted that VoIP not only converges the voice and data services over IP, but it also converges the control and data planes over IP, both of which were previously served by separate networks in the classical voice systems.

3.1.4. Quality of Service Requirements

As a result of history, voice services come from a circuit switched paradigm-where transmission resources were reserved for the duration of the call, as a result users have become accustomed to a certain level of voice quality. However, VoIP services come from the packet switched paradigm. This paradigm did not develop with voice services in mind, as such there is no guarantee that resources will always be available along the path to service an active call, it is still a best effort service. There are a number of metrics that affect the user's quality of experience namely, delay, delay variation/jitter and packet loss. This section describes the details of these metrics, how they affect the quality of experience and the required values for each in order to satisfy user's quality of experience expectation.

Delay refers to end-to-end delay, in other words the time it takes for a speaker's utterance to travel from the mouth to the listener's ear. According to [61] there are five contributions to the overall end-to-end delay namely the encoding delay, packetisation delay, network delay, playback delay and decoding delay. Encoding delay refers to the time required to convert the voice analogue signal into a digital signal and compress it. This depends largely on the chosen codec scheme. The delay contribution of some of the common speech codecs are outlined later. Therefore the best codec is one that is less complicated to implement, least delay, least speech rate and the highest Mean Opinion Score (MOS). The MOS is a test wherein users of a telecommunication system rate the network quality based on subjective tests [62].

However, there is probably no codec with all these characteristic and sometimes it is a case of trial and error to determine the best codec for the prevailing network conditions. As a result the effects of choosing codecs are also considered in this investigation.

The second contributor to end-to-end delay is the packetisation interval. This refers to the amount of time it takes for the encoded speech to be packaged into packets [8]. The shorter the packetisation interval, the faster it takes to packetize the speech and the lower the delay contribution. However, a short packetisation interval reduces payload efficiency [2]. This means the ratio of actual speech data to overhead is lower, thus there is more overhead traffic in the network than there is user data, hence inefficient usage of the bandwidth. On the other hand, the higher the packetisation interval the better the payload and bandwidth utilization efficiency, but this comes at a cost of additional delay which negatively affects the end user experience [2]. Therefore, the choice of a packetisation interval is also a balancing act. In [63] Tobagi *et al* investigated the packetisation interval for G.711, G.729 and G.723. They concluded that a 30ms packetisation interval is a reasonable compromise for G.729 and G.723 codecs to balance payload efficiency and latency. However, they also concluded that for G.711 a packetisation interval of 10ms performed better. In this research, a packetisation interval of 30ms is used.

Network delay refers to the time taken between packets leaving the sender's device to the packet reaching the receiver's device. While propagating through the network, the packet will experience propagation delays between the links. The packet will also experience queuing delays in the intermediate nodes because packet service is a best effort service.

The UDP protocol does not guarantee a sequenced arrival of packets at the destination device. Therefore, packets can arrive out of sequence and playing back packets that are out of sequence can render a VoIP system unattractive to an end user who is accustomed to carrier grade voice services [2]. VoIP systems are therefore designed with a playback buffer. The purpose of the buffer is to ensure that the packets are played in the correct order and the packet sequencing is a service that is provided by the RTP protocol. Therefore packets have to be buffered for a certain amount of time in order to ensure correct playback. This is referred to as the playback delay.

The digital packet stream still needs to be converted back to an analogue audio speech. This conversion process also introduces additional delay that is referred to as the decoding delay. All these delay compound and add up to the end-to-end delay. ITU-T in recommendation G.114 [64] proposes a one way end-to-end delay of below 150ms for terrestrial voice. In exceptional cases such as voice carried over satellite, which has a long round trip delay, the user expects some level of delay and here [64] suggests a figure up to 400ms for acceptable voice communication.

Packet switching is a best effort service, as such packets will arrive at different times depending upon the service they receive along the transmission route. Delay variation also known as jitter refers to the unpredictability in the arrival times of packets at the receiver. Therefore in order to ensure intelligible voice communication VoIP systems implement a buffer at the receiver end to ensure that packets are played back in the correct order. If the buffer is full any arriving packets will be dropped. Albeit addressing the problem of jitter, a buffer can introduce packet losses which are undesirable for voice communications. ETSI TR 101 [65] recommends a jitter value between 30ms and 75ms for an acceptable voice conversation.

UDP protocol that is used to carry VoIP system does not guarantee delivery of packets. As a result any packet lost along the transmission route can never be recovered. A packet can also be discarded if it fails a checksum at the receiver. An intermediate router can drop a packet if the router's buffer is full [61]. A packet may be dropped at the receiver if it arrives outside of the jitter buffer length. In [66] it is reported that a packet loss rate of 2% is tolerable for voice communications. The problem of packet loss is especially pronounced in wireless systems, because of the very unfriendly transmission medium that suffers from multipath fading and interference. Forward Error Correction (FEC) is proposed in wireless systems to minimize packet-loss. However, FEC introduce additional delay, which also degrades voice communications. Therefore, when choosing an FEC technique a balance needs to be struck between additional delay and packet-loss.

3.1.5. Speech Quality Evaluation

During a voice conversation, speech signals experience various forms of degradation such as losses, circuit noise, transmission errors, environmental noise, side-tones, echo, propagation delay [62]. These distortions impact the quality of speech signals and thus the quality of the user's experience in the network. Therefore it is important that tools are available to assess the quality of speech. To this effect, two categories of speech quality methods exist, namely objective and subjective methods.

3.1.5.1. Subjective Speech Assessment

Subjective methods are those that depend on the opinion of the people who participate in the evaluation experiment. ITU-T provides details of the various subjective tests. In essence there are two methods to carry out subjective testing listening only tests and conversational tests [67]. Conversational tests are the preferred tests, but they are difficult to carry-out therefore the most practical methods are the listening methods. Some examples of listening methods include Absolute Category Rating (ACR), Quantal-Response Detectability Method (QRD), Degradation Category Rating (DCR), Comparison Category Rating (CCR) and Threshold method (TR) [62].
ITU-T in [67] recommends ACR method for listening only testing as this conforms to the methods used for conversational tests. Furthermore, this method has been applied to testing different codec schemes across independent laboratories with a great degree of repeatability. In this method the test subject assigns opinion ratings of 1-Bad, 2-Poor, 3-Fair, 4-Good, 5-Excellent are assigned by participants to groups of independent sentences. These scores are referred to as the Mean Opinion Scores (MOS).

The QRD tests are suitable for evaluating detectability of a qualitative property of speech such as echo as a function of a quantitative property such as listening level [66]. In this method, test subjects assign opinion ratings as follows:

- A-objectionable,
- B-detectable,
- C-Not detectable

These ratings are given to the qualitative property at different values of the quantitative property. The scoring allows the assessor to understand the listening level at which echo becomes unacceptable. This method can also be used to assess the level at which a particular impairment such as interference has a higher probability of creating discomfort for the listener.

The DCR is an extension of the ACR, but uses a high quality reference, unlike the ABR where reference is the participant's experience. The participating listener is presented first with the reference sentence, then with the same sentence but slightly degraded through processing. The deviations of the of the test sentence, from the high quality reference are given opinion scores of

- 1-Degradation is very annoying,
- 2-Degradation is annoying,
- 3-degradation is slightly annoying,
- 4-degradation is audible but not annoying,
- 5 –Degradation is inaudible.

The average of these values is referred to as the Degradation MOS [62]. This method is suited for assessing the impact of degradation of speech through the channel, on the quality of user's experience.

The CCR is similar to the DCR, but here the test sentences are presented at random. The second sentence is compared to the first and assigned an opinion scores as follows:

- -3 Much worse,
- -2 Worse,
- -1, slightly worse,
- 0 about the same,
- 1 slightly better,
- 2 better
- 3 Much better.

This method is suitable for assessing the impact of degradation or improvement on the quality of user experience.

3.1.5.2. Objective Speech Assessment

In order to objectively and fairly compare the performance of different communication systems, both objective and subjective methods are used. The subjective are translated through experiments into objective methods. Subjective methods are measured on the average opinion of a selected group of listeners and speakers. The limitations of subjective tools is that in most cases they are not really repeatable [61] and they are also not suitable for assessment of operational systems because they are intrusive, expensive and time consuming [67]. The objective methods seek to estimate subjective opinion scores without going through the costly subjective experiment, ITU-T rec P.861, [68]. This section covers details of the objective techniques namely: Perceptual Evaluation of Speech Quality (PESQ) and Perceptual Speech Quality Measure (PAQM) and finally what may be referred to as a hybrid technique, the E-model. The Figure 32 below from [69]shows the concept of objective measurements. As per figure, the speech is generated and one a copy is processed while the original is kept. The processing simulates any codec scheme, packet loss, delay or jitter. The processed signal and the unprocessed signal are then compared using an objective algorithm. The results are then transformed to a subjective view to give an estimated MOS.

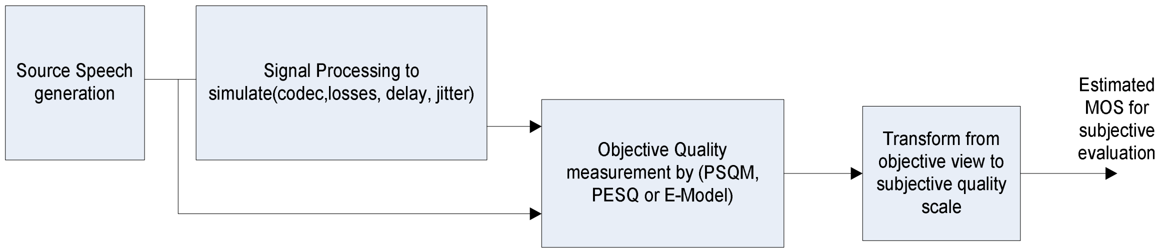


Figure 32: Objective Measurements criteria. **[69]**

Perceptual Speech Quality Measure (PSQM) is specified in [67] as the first attempt to standardize an objective evaluation method on the back of weaknesses in subjective methods highlighted earlier. This method estimates the quality in a similar way to listening only DCR subjective methods. The algorithms for these computations are explained in details in ITU-T recommendation P.862 [70]. It was designed to assess different codecs, therefore was not suitable to assessment of voice degradations in systems. Due to its unsuitability for evaluation of speech in network systems, this specification was withdrawn.

Perceptual Evaluation of Speech Quality (PESQ), ITU-T Rec P.862 replaced the PSQM. PESQ was standardized not only for codec evaluation but also for assessing narrow-band telecommunication systems which could be affected by unpredictable delay, filtering, channel errors and low-bit rate codecs [70]. PESQ addresses these issues through channel equalization, averaging distortions over time and time alignment. In its own admission, ITU-T P.862 states that this method cannot replace the subjective evaluation, as such it is possible for a system to score high on this method and still have a poor voice quality. This is because this method is based on the listening only subjective techniques wherein only the one way speech distortion and noise impairments are measured.

The E-model, ITU-T Recommendation G.107, in [69] was introduced in order to address the short-comings of the above methods, by adopting the conversation methods approach of the subjective evaluation methods. According to the specification document, this model includes transmission impairment factors that mimic the real-world transmission systems. The output of this model is a so called R-factor, see equation (4)below, which can be mapped to a MOS i.e. customer opinion scores from the subjective paradigm. The specification emphasizes that this method is merely for comparison of the transmission conditions and not really reflective of the customer experiences.

$$R = R_o - I_d - I_e - I_s + A \qquad \text{(Equation 3) [69]}$$

-R_o: Captures the essence of Signal to Noise ratio.
- I_s: Captures combination of all the distortions that occur with the voice.
- I_d: Captures Impairments due to delay.
- I_e: Captures Impairments due to low bit rate codecs.
- A: Compensation of impairments factors when there are other advantages of access.

The mapping between MOS and the R-Value is done as per below and in Table 8.

$R < 0$: MOS = 1

$0 < R < 100$: MOS = $1 + 0.035R + R(R-60)(100-R)7*10^{-6}$

$R > 100$: MOS = 4.5

Details of calculations for the components can be found in ITU-T Recommendation G.107, [69]. OPNET uses the R-Value to calculate the MOS.

Table 8: R value to MOS mapping to user satisfaction mapping **[71]**

R-value(e-Model)	MOS	User Satisfaction
90	4.34	Very Satisfied
80	4.03	Satisfied
70	3.6	Some users dissatisfied
60	3.1	Many users dissatisfied
50	2.58	Nearly all users dissatisfied
0	1	Not recommended

3.1.5.3. Speech Codecs

Bandwidth is a scare and expensive resource in telecommunications. Therefore, clever techniques are required to utilize this resource efficiently. Compression/decompression (codec) schemes are an example of the techniques that seek to maximize efficiency of the network bandwidth, while maintaining an acceptable quality of user experience. Human speech is analogue in nature, and it was traditionally carried over analogue networks. However, analogue systems are inherently bandwidth inefficient [61] compared with Digital communications systems. The main advantage of digital systems is that digitized signals can be manipulated and operated upon. One of the operations that can be carried out on the digital signals is the compression and decompression. This is possible because bit patterns repeat. The compression/decompression of voice is performed by speech codecs. These codecs fall into three categories, namely Narrowband, Wideband and Multimode.

The Narrowband codecs operate on narrowband speech i.e. 0-3400 Hz. The Nyquist criterion would dictates that to avoid aliasing, this speech should be sampled at least 7.8 KHz [14]. The actual sampling rate used is 8KHz. Some of the narrowband codecs such are G.711, G.723.1, G.726, G.726A, G.729, G.729A, G.729B. These are specified by the ITU [49] and can be identified with the “G.” prefix. Other codecs are specified by ETSI such as GSM06.20 (GSM half rate), GSM 06.10(GSM Full rate) and Adaptive Multi Rate (AMR).

Wideband voice implies that instead of 0-3.4KHz, the sampling is done over a wider spectrum and includes more high frequencies to increase the quality of speech. In this case frequencies up to 7KHz are included [14]. Therefore, the Nyquist criterion dictates a sampling rate of at least 14 KHz to avoid aliasing. The actual sampling rate used in Wideband is 16 KHz [14]. This section will discuss details of the common codec schemes and also highlight key differences between them.

The G.711 was for the longest time the most popular speech codec because of low computational demand, low delay and low distortion [49]. However, as the computing power became less expensive, it was no longer necessary to live with the weaknesses of this codec, mainly that its data rate was just too high. The Pulse Code Modulation scheme samples the voice signal at 8KHz and uses an 8bit quantization, therefore resulting in 64Kbps (8000 samples per second * 8bits per sample). This system uses a logarithmic companding [61]. The logarithmic companding is implemented differently in Europe and America, namely A-Law and mu Law respectively. The A-law implementation of G.711 compresses 13-bit samples into 8-bit, whereas the mu-Law version compresses 14-bit samples into 8-bits. The details of the two companding schemes can be found in [14] and [72].

The G.723 is another ITU-T codec that is capable of two speech rates, 5.3Kbps and 6.3Kbps. The codec uses the Multi Pulse-Maximum Likelihood Quantization (MP-MLQ) for the 6.3Kbps rate and 5.3Kbps with Algebraic Code Excited Prediction Algorithm (ACELP). This codec was designed for implementation with the H.323 streaming standards [61]. It introduces a 37.7ms algorithmic delay, according to [49]. Therefore, when using this codec 37.7ms of delay is introduced before the packet has even left the sender.

The G.726 is capable of four bit rates, 16, 24, 32 and 40 Kbps and uses Adaptive Differential PCM coding (AD-PCM). AD-PCM saves bandwidth by measuring the deviation of samples from the predicted as opposed to deviation from zero [73]. This results in less bits being required to represent a sample. This codec is an extension of the G.711 and seeks to reduce the 64Kbps to 16, 24, 32 and 40 Kbps. However, [61] contends that the 24 and 16 Kbps result in very poor speech, hence only the 32 and 40 Kbps rates should be considered for real life application.

The ITU-T G.729 uses the Conjugate-Structure Algebraic-code-excited Prediction (CS-ACELP) to encode the speech at 8Kbps but with a fairly high fidelity. This is a fairly complex algorithm; hence it requires more processing power in the device. As a result, ITU-T introduced G.729A which is less complex [61]. The two codecs are interoperable.
The table below compares the most popular speech codecs.

Table 9: Comparison of Narrow band, wideband and multimode codec characteristics [74].

Codec Information				Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth MP or FRF.12 (Kbps)	Bandwidth w/cRTP MP or FRF.12 (Kbps)	Bandwidth Ethernet (Kbps)
G.711 (64 Kbps)	80 Bytes	10 ms	4.1	160 Bytes	20 ms	50	82.8 Kbps	67.6 Kbps	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	3.92	20 Bytes	20 ms	50	26.8 Kbps	11.6 Kbps	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	3.9	24 Bytes	30 ms	33.3	18.9 Kbps	8.8 Kbps	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	3.8	20 Bytes	30 ms	33.3	17.9 Kbps	7.7 Kbps	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	3.85	80 Bytes	20 ms	50	50.8 Kbps	35.6 Kbps	55.2 Kbps
G.726 (24 Kbps)	15 Bytes	5 ms		60 Bytes	20 ms	50	42.8 Kbps	27.6 Kbps	47.2 Kbps
G.728 (16 Kbps)	10 Bytes	5 ms	3.61	60 Bytes	30 ms	33.3	28.5 Kbps	18.4 Kbps	31.5 Kbps
G722_64k(64 Kbps)	80 Bytes	10 ms	4.13	160 Bytes	20 ms	50	82.8 Kbps	67.6Kbps	87.2 Kbps
ilbc_mode_20(15.2Kbps)	38 Bytes	20 ms	NA	38 Bytes	20 ms	50	34.0Kbps	18.8 Kbps	38.4Kbps
ilbc_mode_30(13.33Kbps)	50 Bytes	30 ms	NA	50 Bytes	30 ms	33.3	25.867 Kbps	15.73Kbps	28.8 Kbps

It should be noted though, that in choosing a codec scheme, all the factors; codec rates, delay, complexity of the algorithm and MOS have to be taken into account. The system designer needs to trade-off the different factors in order to provide the best possible quality of experience. In this study the G.711 is chosen as the preferred codec for this investigation. The G.711 codec presents the worst and the best in that it has the highest data rate requirements of all narrow band codecs, i.e. 64Kbps. Furthermore it is the least complex codec scheme to implement. Finally, it gives one of the best MOS values of all narrowband codecs. The high data rate is suitable for this evaluation because it will test the Wi-Fi network's ability to carry heavy data that is generated by this codec. The assumption is that if Wi-Fi can carry speech that is based on this codec with distinction then it should carry speech that is based on any other codec with ease.

3.2. File Transfer Protocol(FTP)

File Transfer Protocol (FTP) is a method that is used to transfer files between systems in a computer network. Unlike VoIP, FTP traffic uses Transport Control Protocol (TCP) to exchange files see the protocol stack in Figure 33 [17]. The figure shows that FTP uses the transport services of TCP. Unlike UDP, TCP is a referred to as a reliable and connection oriented protocol. It is a connection oriented protocol in the sense that an end to end session needs to be established and maintained and every packet sent is acknowledged [17]. The connection state resides in the end systems; therefore the network is unaware of this connection [6]. TCP is also a point to point protocol, although a single end system can establish parallel TCP connection [6]. In this work a single FTP server, establishes parallel FTP connections to several Wi-Fi stations.

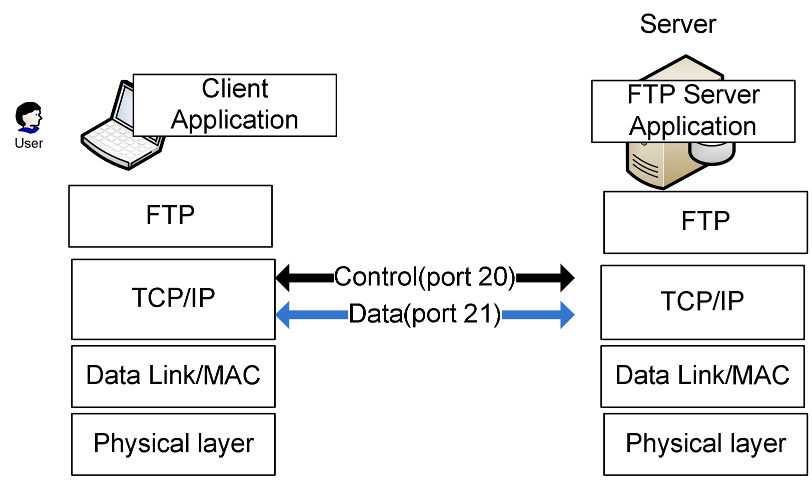


Figure 33: FTP Protocol Stack **[6].**

FTP maintains a separate control and data connections Figure 33, this is referred to as out of band signalling. FTP control connection state is persistent in other words it is established once and it is maintained throughout whereas the data connection needs to be established for each file transfer.
The congestion control nature of FTP uses TCP congestion control mechanisms. These mechanisms ensure throttling of senders if there is excessive packet-loss on the network. TCP throttling halves the data rate of all senders if a packet-loss occurs on the network [6]. This approach to congestion management creates bursty traffic which increases queuing delays, creates unpredictable queues (jitter), leads to packet-loss and reduces throughput [75]. Furthermore, large TCP default windows were optimised for a wired network, which tends to be more forgiving when it comes to errors and data corruption [75], but these are not suitable for wireless networks. An understanding of the effect of these issues on VoIP is critical to achieving a carrier grade VoIP experience. This is because VoIP services utilise UDP which has no flow control, packet-loss recovery or congestion control. In this work we investigate whether in their current form the MAC mechanisms are sufficiently robust to support these UDP weaknesses.

4. Chapter 4: Trends in Voice over IP over WLAN

In earlier sections, the research work and background were introduced. Furthermore, the technical workings of the WLAN technology and VoIP technology were discussed in detail. In order to put this research into perspective, this section reviews some of the earlier work that has been carried out with regards to VoIP over WLAN in the presence of other data.
Most of earlier research has focussed on improving the number of simultaneous calls (capacity) of VoIP over WLAN technologies. In some of the previous work, the investigation of WLAN capacity under the different coordination schemes was completed without regard for other traffic types. This is in spite of the fact that QoS enhancements are only useful if the high priority stations are in a mixed traffic network. Furthermore, previous work has neglected the fact that VoIP over WLAN station does not operate in isolation but they operate alongside of other data applications stations that are competing for the same medium. Moreover, there is major difficulty with designing admission control schemes for VoIP in the presence of other data, mainly because of the difficulty with estimating channel capacity. As a result of the difficulty that comes with trying to estimate the blocking point in a mixed VoIP over WLAN, the researchers in [76] proposed a Channel Utilization Estimate (CUE) metric to determine the network capacity, thus the blocking point, where network capacity is the maximum number of stations that can utilize the channel for various services. Admission control is very critical in VoIP systems because accommodating an extra call beyond the capacity of the system deteriorates all the on-going calls.

Admission control mechanism are more important in VoIP over WLAN because when the call capacity has been reached, granting additional request to the medium degrades all the on-going calls and this can severely compromise the service [77]. The degradation in WLAN systems occurs because when its capacity is exceeded, the coordination schemes mainly DCF, throttles the station with the highest data load, which is the Access Point (AP) in WLANs. In infrastructure mode all the data from all the stations is relayed via the AP, as such the throttling affects all the station in the BSS infrastructure. Therefore is it imperative to understand the performance of the existing coordination schemes and their QoS enhancement schemes in the presence of other data under different loading scenarios, because this understanding will lead to development of robust admission control schemes.

Performance of DCF against EDCF was compared by in [78] for the IEEE 802.11b network. In that work Puschita *et al* simulated performance based on CISCO recommended parameters, also known as default parameters settings in QualNet. They concluded that there is some difficulty in translating the application requirements in to a set of WLAN parameters. Furthermore, their results suggest that EDCF performs better on service differentiation than DCF. However, they make no consideration for a changing network and therefore their results are only applicable to that particular network setup.

In [79] a Markov chain model was used to evaluate the performance of DCF in the absence of hidden stations and transmission errors. The throughput, number of contending stations, and data rate were assessed. They concluded that when using basic access methods, throughput depends on the number of stations. Moreover, they found that the RTS/CTS method does not bring additional advantages when the number of contending stations is small. Furthermore, they considered various loading scenarios on the network and showed that there are situations where enhancement techniques such as RTS/CTS are no better performers than ordinary coordination techniques.

In [80] Masnoon *et al* compared the performance of PCF against DCF, in their investigation; they considered performance of PCF when voice is supported by PCF and data supported by DCF and vice versa. They used the G.711 codec in their investigation. They concluded that PCF results in a more acceptable delay than DCF. However, delay is only one aspect that determines the user experience. They did not investigate the overall experience using MOS. Therefore the performance with respect to jitter and packet loss is not understood. Moreover, it is unclear whether the discrepancy in the performance may have been influenced by the choice of codec scheme or packetisation interval, as suggested in [63].

The effect of data traffic on voice is investigated by amongst others [81]. In their work they used PCF for voice traffic and DCF for data traffic without any other QoS enhancement. They found that by lengthening the Contention Free Period when PCF is in operation, they were able to increase the number of voice stations, but this came at the cost of reduced quality of experience for data services, as measured by the total throughput. In this research the work in [81] is extended by assessing the experience under different ratios of other data to voice stations and also by comparing this performance to that of other coordination schemes.

In article [82], Takehiro *et al* investigated the performance of VoIP stations where there is a minority of FTP stations. They found that their proposed approach called the Dynamic PCF improves the performance of voice services both in delay and throughput for FTP stations. The focus of their work was to propose a new coordination technique as opposed to evaluating the existing schemes. Moreover, they only considered a situation wherein the voice stations dominated the infrastructure. However, in reality the ratio of voice stations to data stations is arbitrary and it is imperative to understand how the existing coordination functions compare.

Article [83] proposed a VoIP capacity model based on PCF. They considered voice quality as the determining factor for the model. Their model suggests a higher VoIP capacity for PCF than DCF coordination schemes for high and medium quality voice. However, their assertions have not been tested in a realistic modelling tool such as OPNET. Moreover, they did not consider the effect of other traffic types which always co-exist with voice traffic.

The Wang *et al* in [84] identified capacity of VoIP over WLAN in the presence of other TCP traffic as a major stumbling block for VoIP over WLAN. According to their calculations an 11Mbps WLAN access point should be able to support 550 VoIP sessions. However, this is not the case because of the overheads from RTP, UDP, IP, MAC and Physical layer protocols in the stack. Therefore, they propose a method that aggregates and multicast different IP streams in order to reduce the IP overheads. They report a 100% capacity improvement through this technique. However, this is only an analytical model; no results were reported for simulation or implementation.

In article [85] Wu *et al* studied the voice capacity of WLANs in the presence of data. They identify the challenges of dealing with voice traffic in that, voice does not always generate traffic and also the voice packets are smaller than data packets. As a result of the 40byte voice overhead, the efficiency of voice packets is highly reduced. They identify three previously applied techniques to improve performance of VoIP services over WLAN in the presence of other data namely: MAC header compression; Frame aggregation and differentiated services that give voice traffic priority on the channel. They propose a teletraffic queuing model for IEEE 802.11e based VoIP services to improve capacity. They report a VoIP capacity of 16 stations when there are 6 data stations, 18 VoIP stations when there are 4 data stations and 22 VoIP station where there are 2 data station. Beyond these values the delay for VoIP stations deteriorates quickly.

In article [86] Farooq *et al* investigated the capacity of IEEE 802.11b DCF mode in the presence of background traffic on a test-bed. They tested capacity with Back-off with Prioritized queuing (BC-PQ) and with BC-PQ at the Access Point (AP). The BC-PQ mechanism aims to prioritize the traffic based on some criteria. They tested under heavy background traffic and under light background traffic. They then used a maximum packet-loss of 2% as their performance metric. They concluded that by using BC-PQ under light background traffic conditions, the IEEE 802.11b can accommodate 8 live calls before the packet loss becomes unacceptable compared with 5 calls without BC-PQ. Their second conclusion was that under heavy background traffic conditions with BC-PQ, 8 simultaneous calls can be connected compared to 0 calls without the BC-PQ enhancement. However, their approach only considered 1 codec scheme G.711 with silence suppression and the packetisation interval of 10ms. Therefore it is possible that with a different codec scheme and a different packetisation interval, this performance can be improved upon.

In article [87], Lucani *et al* propose a fair MAC layer algorithm to improve the capacity of the 802.11 WLAN networks. The basis of the algorithm is that, under infrastructure mode with DCF in operation, the AP has the same probabilistic access to the medium as all other stations. They argue that this is flawed because; in this mode the AP has to support all the mobile stations in the BSS. Therefore it makes sense that the AP gets a higher priority access or else it becomes the bottle-neck. They conclude that under low background traffic, the fair algorithm increases the capacity of VoIP users. However, as the background traffic increases, this advantage is lost. PCF inherently gives the AP control over the network as it is the polling scheduler.

In [88], Brouzioutis *et al* propose a mathematical model for computing the voice performance metrics in the presence of data. They predict a linear decrease in voice capacity of 2 voice sessions per data stream. They also propose RTS/CTS mechanism for voice as it will introduce predictability in the behaviour of voice stations. This is further evidence of the need to thoroughly investigate PCF as a medium access mechanism for voice in the presence of other data.

In [89] Medapalli *et al* investigated the VoIP capacity of IEEE 802.11b, a, and g. They concluded that an IEEE 802.11g network can handle about 60 calls with a 10ms packetisation interval. In this work their conclusion about the VoIP capacity of 802.11g is used to decide the number of stations in the BSS.

Table 10: Literature Summary Review

Authors	Contribution
Chatzimisios, P; Boucouvalas, A.C; Vitsas, V	Throughput and Delay Analysis of IEEE 802.11 protocol
Masnoon, Thanthr, Pendse, Rasheed	PCF vs DCF: A performance review
Li, Changle; Li, Jiandong; Cai, Xuelian	Performance Evaluation of IEEE 802.11 WLAN - high Speed Packet Wireless Data Network for Supporting Voice Service
Kawata, Takehiro; Sangho , Shin ; Forte, Andrea	Using dynamic PCF to improve the capacity for VoIP traffic in IEEE 802.11 Networks
A. R. Siddique, J Kamruzzaman	VoIP Capacity over PCF with Imperfect Channel
Wang, Wei ; Liew, Soung Chang ; Li, Victo	Solutions to performance problems in VoIP over 802.11 Wireless LAN
Wu, Yiqun ; Zhu, Yanfeng; Niu, Zhisheng; Zhu, Jing	Capacity Planning for Voice/Data Traffic in IEEE 802.11e Based Wireless LANs
Farooq; Elaoud; Famolari; Abhrajit; Ravichander; Dutta; Prathima; Kodama; Yasuhiro	Voice Performance in WLAN Networks—An Experimental Study
Lucani, Daniel; Badra, Renny; Bianchi, and Carlos	Increasing VoIP capacity on Wi-Fi networks through the use of the FAIR algorithm for MAC
Brouzioutis; Vitsas; Chatzimisios	The Impact of Data Traffic on Voice
Medepalli,Kamesh; Gopalakrishnan, Praveen ; Famolari, David; Kodama, Toshikazu	Voice Capacity of IEEE 802.11b, 802.11a and 802.11g Wireless LANs

The contribution of this work is as follows:

- The effect of FTP on VoIP performance when DCF, EDCF and PCF are used.
- The effect, on PCF performance, of increasing the Contention Free Period (CFP) while only VoIP stations participate in the CFP.
- Effect of introducing fragmentation on both VoIP and FTP traffic.
- The effect of increasing the buffer size of the Access Point on both FTP and VoIP.

5. Simulation

5.1. OPNET Software

In order to evaluate the performance of each of the medium access techniques, simulation was the most practical approach. In choosing simulation environment, several simulation tools were available such as Network Simulator (NS), QualNet and OPNET (**Op**timized **N**etwork **E**ngineering) Tools. However, only OPNET and NS are freely available for educational and non-profit research purposes. Between OPNET and NS, OPNET was found to be the most intuitive, easy to use and interactive tool, hence it was chosen for this research. This section gives a short background to OPNET, the chosen simulation tool for this work.

OPNET is a leading telecommunications modelling software used both by industry and academia. It is a tool for design, analysis of performance; as well as behaviour of networks, protocols and applications. The advantages of software based modelling are that it allows one to study, analyse and understand different scenarios, without the added cost of acquiring the actual hardware [90]. OPNET uses object oriented modelling and a user friendly graphic user interface, both these make OPNET easy to use and intuitive modelling tool. It supports all major network types and technologies, which allow one to design, model and simulate different scenarios with reasonable certainty. Furthermore, OPNET models are a realistic [91] and it can be adapted by changing the C++ code.

A typical simulation process flow is as per Figure 34 below.

- Firstly, one builds the desired network by selecting the relevant network elements and joining them.
- The next step is to choose statistics that are of interest to the investigation. The number of chosen statistics for computation will inherently affect the simulation time. Then the model is ready for simulation.
- On simulation, the user can choose the duration of the simulation. If the network has a lot of nodes and a lot of statistics, is better to choose a shorter simulation time. Unfortunately, once the simulation completes, it is not possible to re-compute the statistics that were not initially chosen. If further statistics are required, an additional simulation needs to be run.

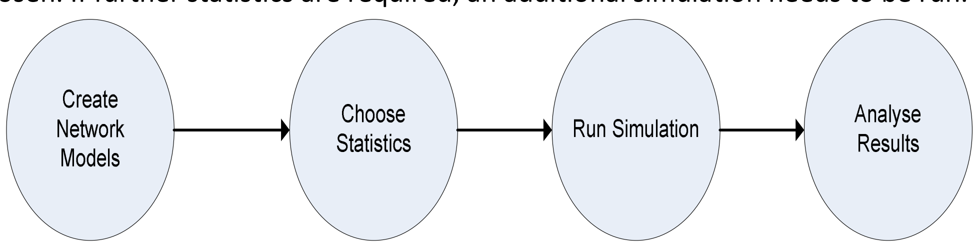


Figure 34: OPNET modelling process (from [51])

OPNET simulation is hierarchical, see Figure 35:

- At the highest level is the network domain which consists of nodes. This domain masks OPNET's complex functions via a simple GUI, while allowing the user to quickly identify his/her desired network nodes. The devices and group of devices are represented by the nodes, examples being workstations, IP-phones, servers, clouds, routers etc. The network level masks the function of the node level see Figure 34.
- The second level is the node level. This level captures the internal functions of a node. Typically they node is modelled using the Open System Interconnect reference model.
- The process level outlines state transitions

- The code level implements the state transition in C++ code.

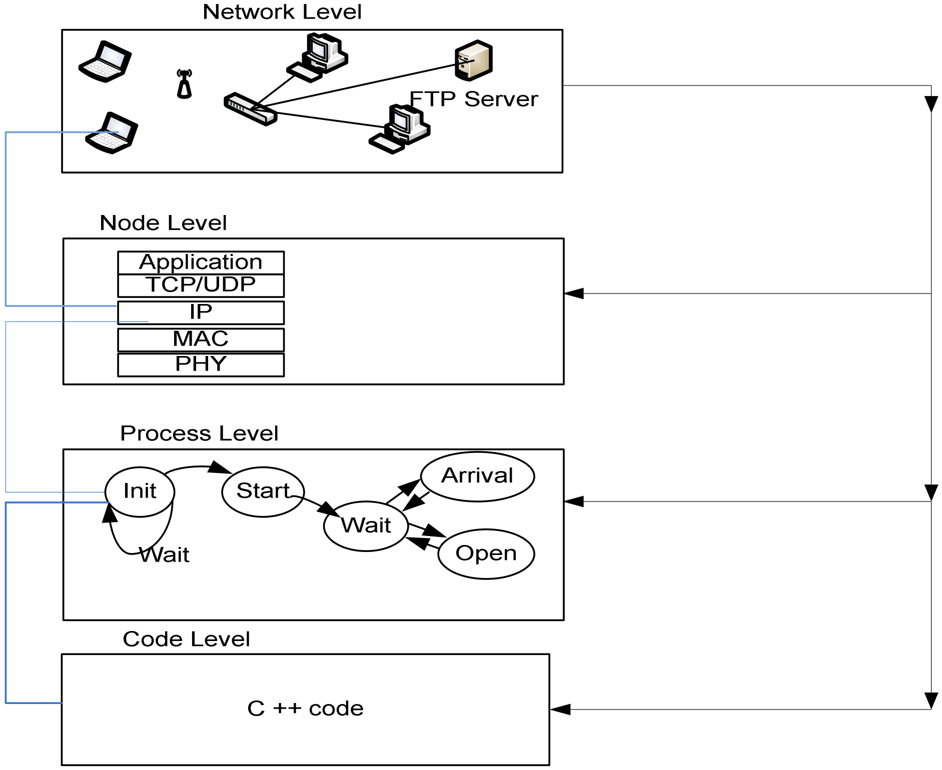


Figure 35: OPNET modelling hierarchy. From [91]

5.2. Simulation

5.2.1. Research Questions

The research question that was poised in section 1.6.2 is answered by investigating the following specific questions:

- With respect to carrying VoIP traffic and FTP traffic with different load ratios, is there a significant difference in performance between DCF, PCF and EDCF?
- How does PCF perform when only VoIP stations participate in the Contention Free Period (CFP) and the CFP is lengthened?
- Does layer-2 breakdown of frames (fragmentation) improve performance of DCF, PCF and EDCF in mixed traffic networks? Does the ratio of data to VoIP station affect performance?
- Does increasing the buffer length when fragmentation is employed improve performance?
- Does the load difference between FTP and VoIP matter?
- Can a lower data rate codec improve VoIP performance for any scheme?

5.2.2. Setup

The simulations were run in OPNET over 30-minute duration. This duration is sufficient for assessing the network behaviour. Simulation durations that are longer than 30 minutes can consume a lot of memory on the simulating machine, thus causing the machine to crash. Although the desire was to run these simulations in the IEEE 802.11n environment, this was not possible because the freely available OPNET software does not support this standard.

The simulations were run for the IEEE 802.11g standard with the following parameters: Orthogonal Frequency Division Multiple Access, data rate of 54Mbps, the default 22MHz channel bandwidth, beacon interval 0.02s, packetisation interval 10ms. They were run with a total of 60-stations with the G.711, see Figure 36. The 60 stations were chosen because this number represents the most practical number of VoIP stations that an IEEE 802.11g network can accommodate [89]. Each station was paired to a peer station in the fixed network. Although the pairing could be done in the Wi-Fi, this approach was chosen in order to simulate a typical scenario whereby a user on a Wi-Fi enabled device receives a call from elsewhere on the internet. Therefore the data from all the WLAN/mobile stations would be relayed via the Access Point through a switch to the peer station in the fixed network. The FTP station all downloaded data from an FTP server.

According to Table 9 the G.711 codec gives one of the highest MOS values, the lowest delay and the lowest computational complexity. This codec also gives the highest data rate which is critical in lossy Wi-Fi networks. Therefore this investigation is carried out on the G.711 which can be considered the worst case scenario for data rate requirements by any codec.

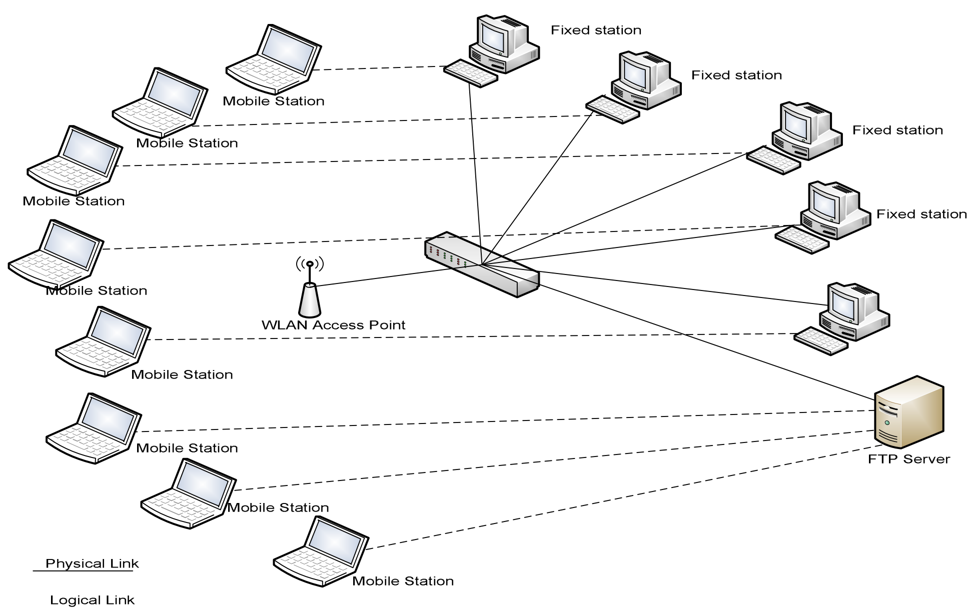


Figure 36: WLAN Implementation

The ratios of the stations were as follows:

- 30-stations generating VoIP traffic while the other 30 stations generate heavy FTP traffic (50% - 50%).
- 15-stations generating VoIP traffic while the other 45 stations generate heavy FTP traffic (25% - 75%).

By default DCF and EDCF all treat the Access Point (AP) as just another station; therefore it is allocated the same probability of gaining access to the medium as any other station. This treatment is however incorrect because the Access Point has to relay the data of many other mobile stations, therefore it ought to have greater access compared to other stations. In [89] it was identified as a potential bottleneck. In light of this observation, we extend the investigation by assigning the AP a longer buffer compared with other stations for DCF and EDCF to assess whether giving the AP greater buffer capacity improves its ability to relay FTP and VoIP.

The above investigation is extended further by providing voice stations a longer CFP compared to FTP stations. This scenario is investigated for a case wherein there is an equal split between VoIP station and FTP stations (30-30) and again in the case where FTP stations dominate the network (45-15). These scenarios are chosen because in a fair BSS a greater number of FTP stations reduce the chance of a VoIP station gaining access to the medium, thus may negatively affect the performance of VoIP traffic.

The e-Model uses packet loss as one of the input factors when calculating the MOS value. A long buffer size reduces the amount of packet loss in the network, but it introduces additional delay. This is especially undesirable for real-time services such as VoIP. Therefore the choice of a buffer length is a critical success factor for the coordination schemes. As a result of this key consideration, we extend the above investigation to the buffer size. We investigate a buffer length that can improve performance of VoIP services without severely affecting the FTP traffic.

Table 11 Summary of simulations

Experiment number	Simulation	number of Stations	Codec	Measured VoIP metrics	Measured FTP metrics
1	PCF vs DCF vs EDCF	30 Voice/ 30 FTP	G.711	end Delay, Voice Packet Delay Variation	response time, FTP traffic throughput
2	PCF performance with lengthened CFP and excluding of FTP Stations during CFP. Compare to PCF in experiment 1	30 Voice/ 30 FTP	G.711	MOS, End to end Delay, Voice Packet Delay Variation	Download response time, FTP traffic throughput
3	Introducing fragmentation of 1024KB & 256KB to experiment 1 above. In each case assess if PCF, DCF or EDCF Voice/FTP performance in experiement 1 can improved by using fragmentation	30 Voice/ 30 FTP	G.711	MOS, End to end Delay, Voice Packet Delay Variation	Download response time, FTP traffic throughput
4	repeat experiment- 1 but increase the Access Point buffer size and compare results to experiment 1		G.711	MOS, End to end Delay, Voice Packet Delay Variation	Download response time, FTP traffic throughput
5	Repeat experiement 1, 2, 3 and change compasition of Voice/FTP stations	15 Voice/45 FTP	G.711	MOS, End to end Delay, Voice Packet Delay Variation	Download response time, FTP traffic throughput
6	Repeat experiement 1 with a different codec and compare the results to experiement 1	30 Voice/ 30 FTP	G.723	MOS, End to end Delay, Voice Packet Delay Variation	Download response time, FTP traffic throughput

5.2.3. Codec Parameter Settings

The network evaluation was carried out with the G.711 codec. This codec has the highest data rate requirements of 64Kbps and it is the least complex of all the VoIP codec schemes. Therefore, it represents the worst case scenario in term of data rate requirements. Given the lossy nature of Wi-Fi systems, a lower data rate codec is preferred; therefore if the system is evaluated with this codec one can draw inference about the performance of other codecs which require a lesser data rate.

The following parameters were set for the G.711 codec evaluation:

- Packetisation interval: 10ms
- Coding rate 64 Kbps
- Speech activity Detection: Disabled

Codec Parameter for G.723

- Packetisation: 30ms
- Coding rate: 5.3 K
- Speech Activity detection: Disabled

The voice traffic was generated with the following parameters:

- Silence length: none.
    - This means that a Continuous Voice Bit rate was used, i.e. voice traffic was generated continuously, although in reality there are silence intervals.
    - Signalling: SIP
    - 1 VoIP frame per packet.
    - Type of Service: Interactive voice.

The FTP traffic was generated with the following parameters:

- Inter-Request time: Exponentially distributed with a mean value of 360 seconds.
- File Size: constant 50 Kb
- Type of service: Best Effort.

The WLAN was setup with the following parameters:

- Physical Layer: IEEE 802.11g
- Data Rate: 54 Mbps
- Transmit power(Default): 5mW
- Access Point Beacon Interval: 0.02s
- Buffer Size: 64Kbps.
    - Larger buffer sizes were also evaluated.
- Large packet Processing: Drop
    - Packet fragmentation was also evaluated.

The parameters in Table 12 are set by default for EDCF:

Table 12: EDCA TXOP for IEEE 802.11e implementation from [4].

Access Class	Cwmin	Cwmax	AIFS Number(number of Timeslot of 2us each	AIFS(SIFS + number of Timeslots*Duration of Timeslots)	TXOP 802.11a/g/n
AC_VO	3	7	2	50	1.504ms
AC_VI	3	7	2	50	3.008 ms 0
AC_BE	15	1023	3	70	0
AC_BK	15	1023	7	150	0

- Slot = 9µs
- SIFS =16µs
- PIFS = 25µs
- DIFS = 34µs
- AIFS >=34µs (This value is small for high User Priority services and large for low User Priority)

5.2.4. Evaluation Metrics

The following metrics were measured in order to compare the performance of each of the three coordination schemes in order to evaluate the quality of VoIP experience:

- Mean Opinion Score (MOS): OPNET calculates MOS by means of R-value as per (3.1.5.2). The acceptable value for MOS is usually 3.5 [66].
- Packet delay Variation 2% [66].
- End to end delay: The generally accepted value for end to end delay is 150ms in one direction [66].
- OPNET averages these values for all stations in the BSS and presents a final average value.

The following metrics are compared between the three coordination schemes in order to evaluate the quality of FTP experience:

- Download response Time.
- Throughput of received traffic in bytes per second.

Assumptions:

- Assume the nodes are static.
- Assume that there are no adjacent Access Point, therefore the no co-channel interference. This is because the purpose of this investigation is to assess MAC technologies.
- The 1Gbps link was used in the wired LAN, therefore it is assumed that most of the delays are a result of WLAN access.

6. Chapter 6: Results

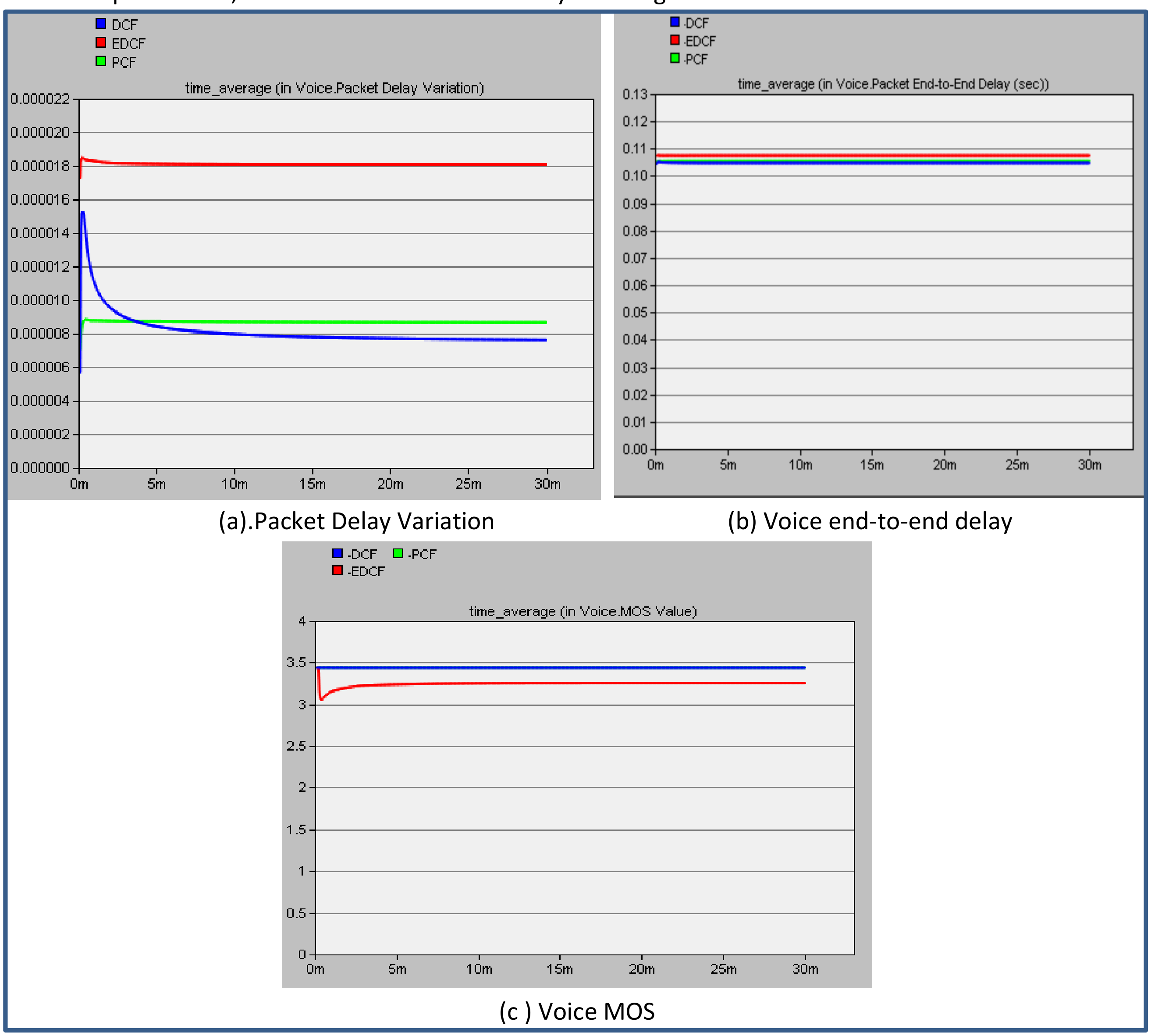
6.1. 30 Voice Stations and 30 FTP Stations with G.711 codec.

In this simulation, the Wi-Fi network consists of an equal distribution (50%/50%) of stations performing VoIP and those that are performing FTP sessions. There are in total 30 FTP stations and 30 VoIP stations. The simulation setup is such that the FTP traffic is largely in one direction, towards the Wi-Fi stations, whereas the VoIP traffic is bi-directional.

6.1.1. PCF, DCF, and EDCF Performance Evaluation.

6.1.1.1. VoIP performance

Figure 37 below outlines the performance of voice services under condition outline Figure 37 (a) indicates that DCF outperforms EDCF with respect to end to end delay variation. Figure 37(b) shows that DCF gives the best end to end delay, followed by PCF then EDCF. Lastly, Figure 37(c), the MOS achieved by PCF and DCF are equal at 3.49, whereas the MOS achieved by EDCF lags below 3.49.



(a).Packet Delay Variation (b) Voice end-to-end delay

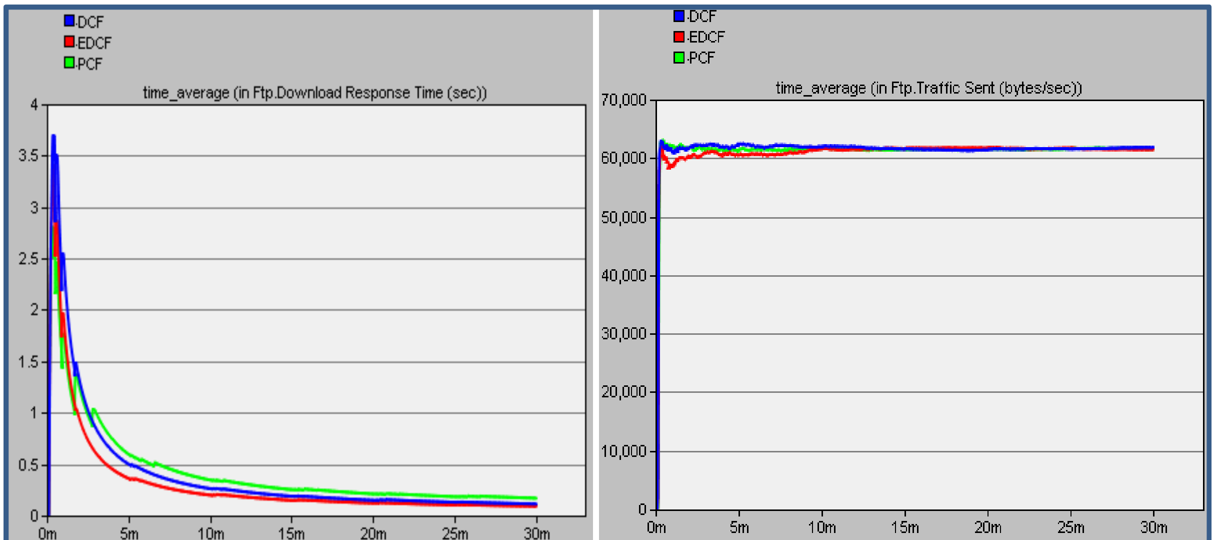
(c) Voice MOS

Figure 37: VoIP traffic experience across different coordination schemes.

These results suggest that in a network where there is an equal number of Voice and FTP traffic stations, the best VoIP performance is achieved if stations are allowed to compete fairly for access to the medium through DCF. The results also suggest that introducing a point coordinator will introduce additional delay variation (jitter), (Figure 37(a)) and will also introduce additional end-to-end delay, Figure 37(b). Furthermore, the results confirm that in an equally distributed network, the current Quality of Service scheme (EDCF) brings no value; in fact it is to the detriment of VoIP performance.

6.1.1.2. FTP Performance

The performance of FTP across the coordination schemes is evaluated using the FTP traffic throughput or average FTP traffic sent in bytes per second. The Download response time gives us an indication of how quickly the FTP traffic is allowed to pass on the downlink.



(a).FTP Download Response Time (b) FTP Traffic Sent

Figure 38: FTP Traffic experience across different coordination schemes.

According to Figure 38(a) PCF outperforms both DCF and EDCF and DCF outperforms EDCF with respect to FTP download response time. This means that an FTP station has a higher likelihood of gaining access to the medium when PCF is in use compared to when either DCF or EDCF is in use. The diagram also means that an FTP station has a higher likelihood of gaining access to the medium when DCF is in use compared to when EDCF is in use. Although the difference in performance seems minute in the figure, in reality, a micro second difference is significant. It is also worth noting that the PCF setting in this case was such that all VoIP and all FTP stations had the same Contention Free Period during which to transmit uninterrupted.

The results suggest that when the Wi-Fi stations are allowed to content for medium access via DCF and EDCF, they have a lower likelihood of gaining access to the medium. Whereas, a better performance is achieved when there is a central coordinator (PCF) dictating how each station accesses the medium.

Figure 38(b) suggests that in the steady-state there is no notable difference in terms of the FTP traffic sent in bytes per second, all schemes give approximately 62Kbps. Therefore from a throughput point of view there is no notable difference between the three coordination schemes.

6.1.1.3. Analysis

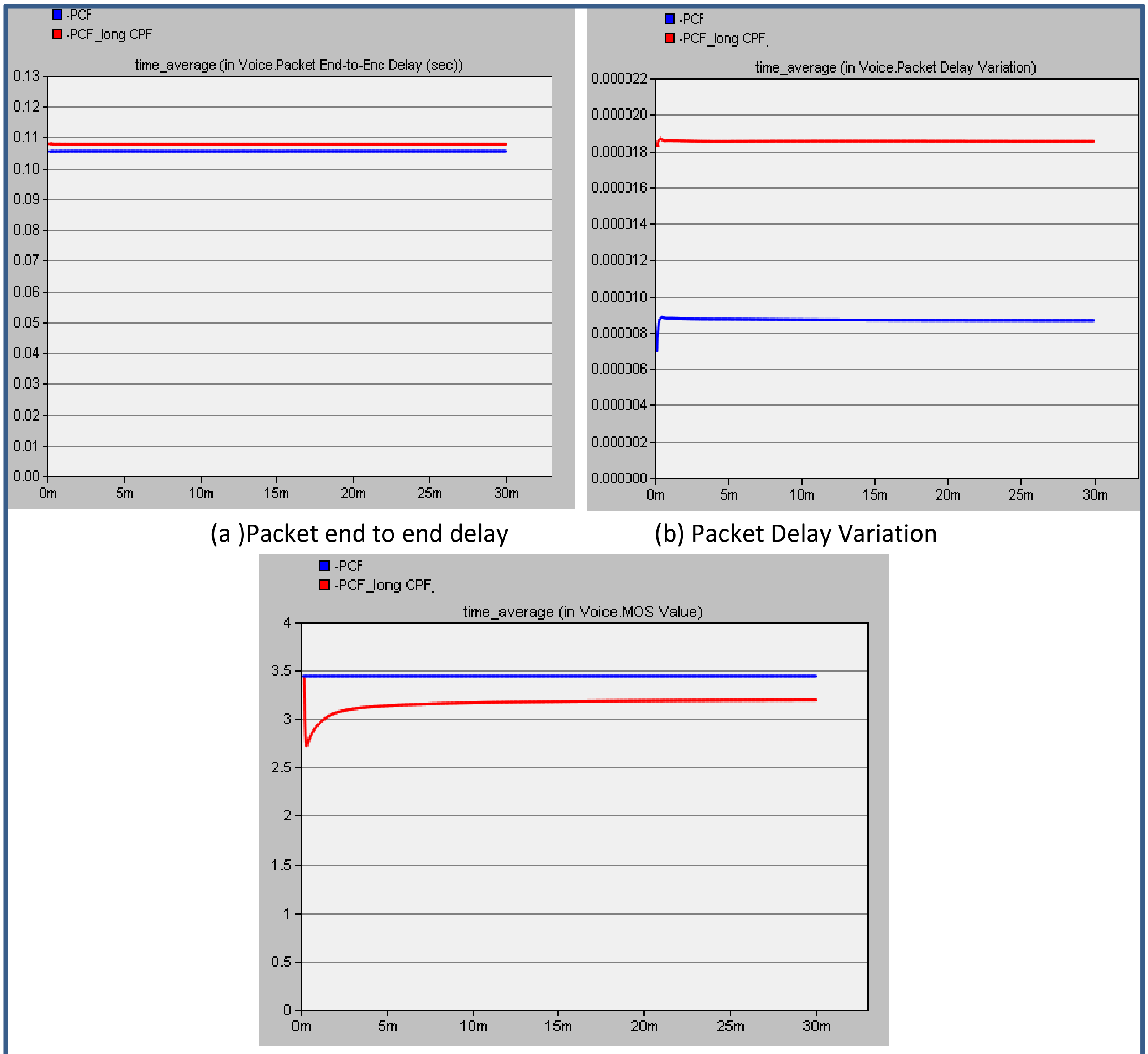
For an FTP user downloading a file, the overall experience is measured by how long it takes to download the overall file as opposed to how long it takes each segment to pass through the network. Therefore, there is essentially no difference in FTP experience across the coordination schemes. For a VoIP user the DCF and PCF give the best voice experience as measured by the overall MOS. The QoS prioritisation of VoIP data by means of EDCF seems to introduce additional jitter and this jitter is responsible for the poor performance of this coordination scheme. Most of the FTP traffic is generated on the downlink towards the Wi-Fi stations. This means that the Access Point is essentially the bottleneck in the system on the downlink. EDCF introduces an additional requirement on the Access Point to examine the header of every frame for Access Class priority before forwarding the frame to the physical layer for transmission. This additional requirement may explain why EDCF suffers a higher end to end delay and a higher delay variation hence a poor voice performance. There is no such requirement for both DCF and EDCF.

6.1.2. PCF Lengthening the CFP while only VoIP Stations Participate.

In order to understand the performance of PCF further, in this section the configuration of PFC was modified so that the duration of the Contention Free period(CFP) was increased from 0.01s to 0.04s and only voice stations were allowed to participate during the CPF. The performance of PCF under the modified configuration was then compared to 6.1.1. This section presents the results of these enhancements. The idea here is to assess whether voice stations can be better served if they have reserved rights to transmit during the periodic CPF while they compete with FTP stations for access during the Contention Period.

6.1.2.1. VoIP Performance

Figure 39 outlines the experience of VoIP stations when the CFP is increased while only VoIP stations are allowed to transmit/receive during the CFP. Figure 39(a) shows an increased end to end delay to VoIP stations as a result of elongated CFP. Figure 39(b) shows a severely degraded packet delay variation and as expected Figure 39(c) shows a degraded MOS from 3.49 to 3.25.



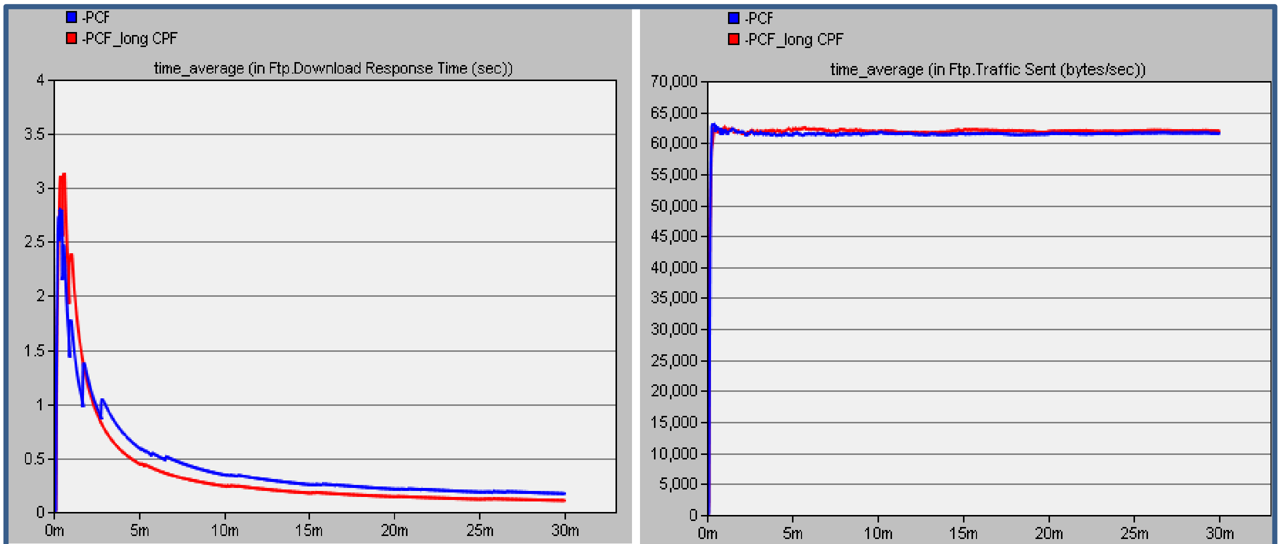
(a)Packet end to end delay (b) Packet Delay Variation

(c) MOS

Figure 39: Performance of VoIP

6.1.2.2. FTP Performance.

The experience of FTP traffic is presented in Figure 40(a) and (b). Figure 40(a) shows that the FTP download response time was reduced when the CFP was increased and only voice station allowed to participate. Although the improvement in FTP throughput is well pronounced, a slight improvement in FTP traffic throughput can be observed in Figure 40(b), i.e. the red line is slightly above the blue line.



(a)FTP Download response. (b) FTP traffic sent

Figure 40: FTP performance for enhanced PCF

6.1.2.3. Analysis

The Contention Free Period allows the Point Coordinator to take control of the medium and in a round robin fashion polls each PCF participant and grants them exclusive rights to the medium. A high-level inspection of these methods may suggest that if Voice stations are granted exclusive access to the CFP and this period is lengthened, this will improve the voice experience. However, an investigation shows that this approach degrades the packet delay variation and end to end delay, hence leading to a poor voice experience as measured by the MOS. During the CFP the medium is held ransom by the Point Coordinator, as a result no station is allowed to transmit and as a result packets continue to build up in every station's buffer until the station is granted access to release the frames. Furthermore, in this investigation the voice traffic was modelled such that there are no silent intervals therefore voice traffic is generated continuously. When packets build up beyond the buffer length of 64 Kb, the station starts to drop some of the incoming packets. This results in an increased packet-loss and hence reduces the overall throughput. The impact is more severe for VoIP station because they are continuously generating packets.

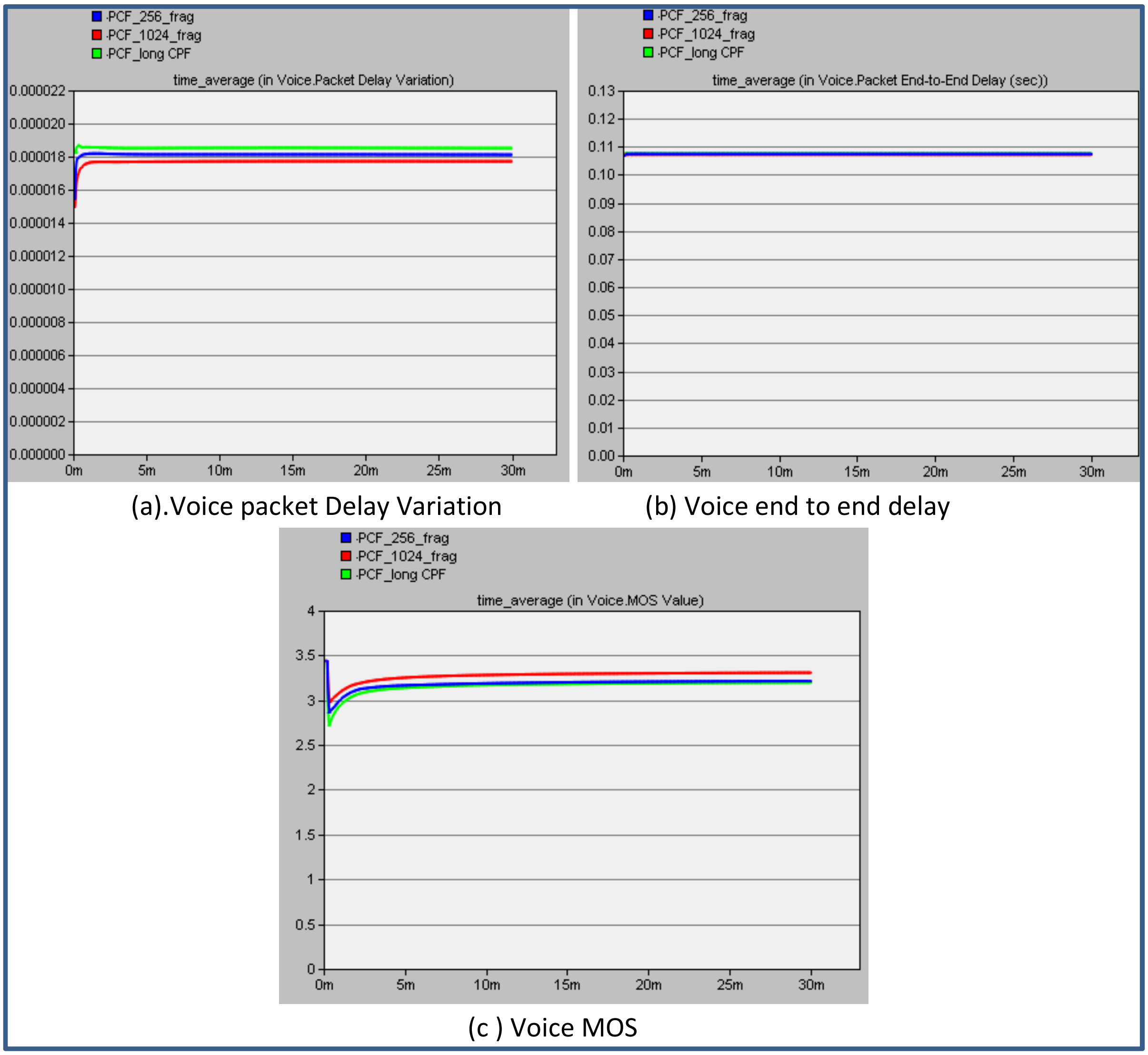
6.1.3. Effect of Fragmentation on DCF, PCF and EDCF performance.

In the previous analysis, frame fragmentation was not used; as a result large packets were transmitted. As was highlighted in section 2.10, the problem with excluding fragmentation is that large packets are sometimes transmitted through the air. If the packet is corrupted, then a large packet has to be retransmitted. This results in inefficient usage of the channel. The problem is especially severe in a lossy wireless medium. The downside of fragmentation is that when a large packet is fragmented, then every fragment has to have its own header, this results in excessive utilisation of the wireless medium to transmit overhead. Furthermore, processing of large FTP packets through the queue can introduce additional delay. In this section the three coordination schemes are re-evaluated individually in order to assess the impact of employing fragmentation with each of the coordination schemes when there is an equal number of VoIP and FTP stations. The effect of fragment sizes of 256 Kb and 1024 Kb on the VoIP and FTP experience is evaluated for each of the coordination schemes. Although the optimal fragment may lie in between the two values, the values were chosen in order to assess the effect of a relatively small fragment versus a large fragment. For each scheme the performance is compared to 6.1.1 to assess if fragmentation brings any additional improvement to any of the schemes.

6.1.3.1. PCF with 256 KB and 1024 KB fragmentation.

i. VoIP Performance

Figure 41 outlines the performance of PCF when fragmentation with fragment sizes 256 KB and 1024 KB are used against the performance when no fragmentation is in use. The figure shows that fragmentation size of 1024 KB gives an improved packet delay variation over fragment size of 256 KB and elongated CFP period, Figure 41(a). As a result it also gives an improved MOS over these two methods, Figure 41(c). However, it provides no improvement to the initial implementation of PCF wherein no fragmentation is used and all stations participate during the CFP with a default, equal and comparatively short CFP.



(a).Voice packet Delay Variation (b) Voice end to end delay

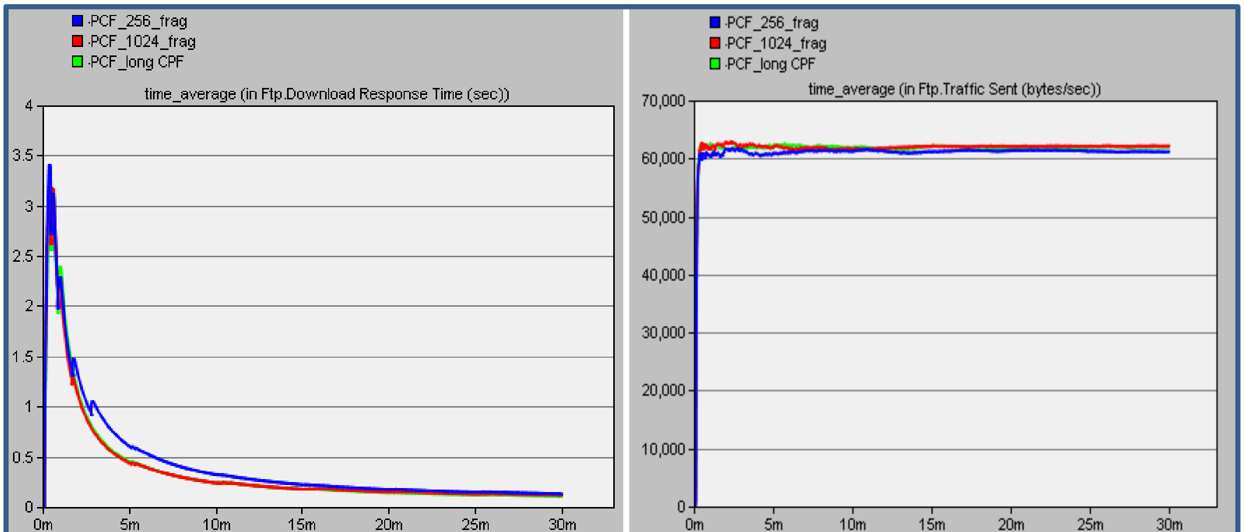
(c) Voice MOS

Figure 41: Effects of fragmentation on VoIP when PCF is used.

ii. FTP Performance

The results below outline the effect of introducing fragmentation when PCF is in use. Figure 42(a) highlights the FTP download response time and it compares PCF with no fragmentation with PCF and fragmentation of 1024 Kb and 256 KB. The results suggest that fragmentation brings no additional improvement to FTP download response time.

In addition, the FTP traffic throughput was assessed when fragmentation is employed and the results are highlighted in Figure 42(b).The results indicate no improvement or degradation in FTP traffic throughput. These results suggest that PCF on its own is effective at managing access to the medium when there are an equal number of VoIP and FTP stations accessing the medium. As such there is no excessive packet-loss, which would usually result from collisions if the medium access is not effectively managed. If there was excessive packet-loss, fragmentation would address this issue, hence there would be visible improvement in FTP throughput.



(a)FTP download response Time (b) FTP traffic throughput

Figure 42: Effects of Fragmentation on FTP when PCF is used.

6.1.3.2. EDCF with 256 KB and 1024 KB fragmentation.

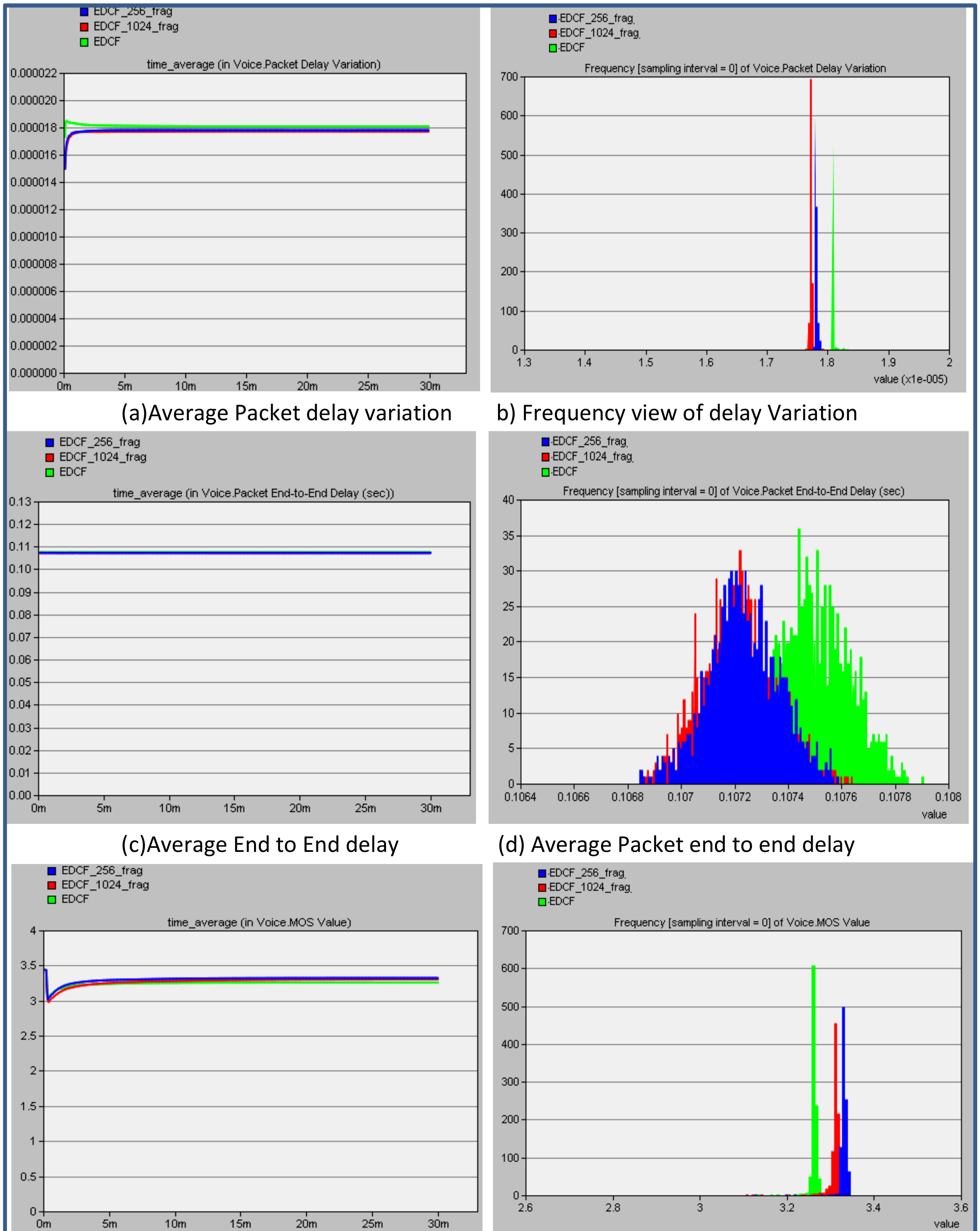
In this section the effect of fragmentation, with 256 KB and 1024 KB fragments on the performance of EDCF for carrying VoIP and FTP traffic simultaneously, was evaluated. A different view of the results, using the frequency plot in included because the average plot does not clearly highlight notable differences. This section also highlights the effect for both VoIP and FTP traffic.

i. VoIP Performance

The Figure 43(a) – (f) presents the effect of employing fragmentation when EDCF is in use. Figure 43(a) and (b) is the packet delay variation. Figure 43(a) is the average value for packet delay variation. The effect of fragmentation is not immediately clear from this plot. An alternative view is the frequency presentation in Figure 43 (b). From the frequency representation, it is immediately clear that by introducing fragmentation in the EDCF environment, the delay variation improved. The larger the size of a fragment the better the packet delay variation.

A similar average view and frequency view is presented in Figure 43(c) and (d), respectively, for the packet end to end delay. Although it is not obvious from the average curve Figure 43(c), the frequency view once again highlights the fact that fragmentation improves the packet end to end delay for VoIP traffic. However, the size of the fragments does not seem to improve performance, because the figure shows EDCF with 256 KB fragmentation and with 1024 KB fragmentation overlay with a center of 0.1072s compared with the EDCF without fragmentation which is centered around 0.1076 seconds.

Figure 43(e) and (f) shows the overall voice experience (MOS) when EDCF is supplemented with fragmentation. Again the average curve does not highlight this effect. However, a frequency plot shows that indeed the MOS improved when fragmentation was introduced. Of interest though is the fact that when the fragment size was increased from 256 KB to 1024 KB the MOS value was reduced.

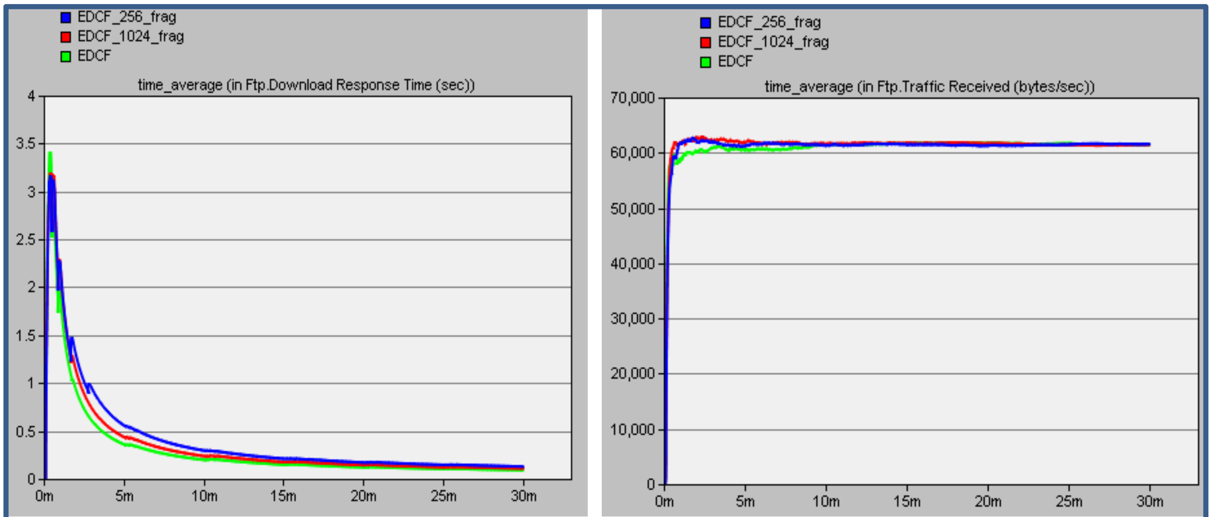


(a)Average Packet delay variation
b) Frequency view of delay Variation
(c)Average End to End delay
(d) Average Packet end to end delay
(e)Average MOS
(f) Frequency Sampling of MOS

Figure 43: Effect of fragmentation on performance of EDCF for supporting VoIP.

ii. FTP Performance

Figure 44 shows how introducing fragmentation to a mixed network that is supported by EDCF affects support for FTP traffic. Figure 44 (a) outlines this effect on the FTP download response time and from the figure it appears fragmentation does not affect FTP download response time. Figure 44 (b) outlines the FTP traffic throughput in bytes/sec. This figure also suggests that fragmentation has no effect on FTP traffic in a 50/50 mixed network, when EDCF is used.



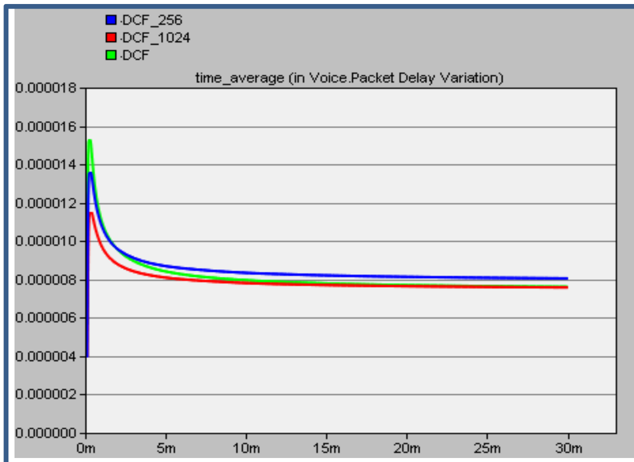
(a)FTP download response time (b)FTP traffic throughput

Figure 44: Effect of fragmentation of on EDCF support for FTP traffic

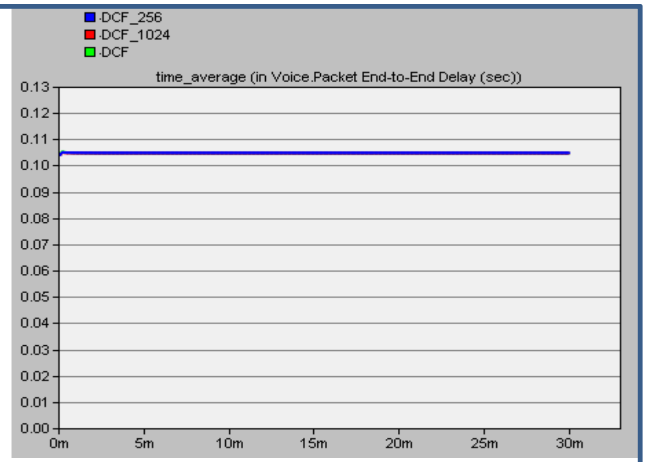
6.1.3.3. DCF with 256 KB and 1024 KB fragmentation.

i. VoIP Performance

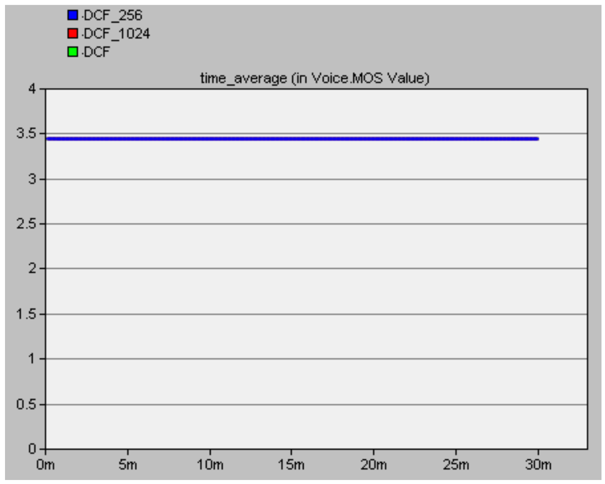
Figure 45 shows the results of VoIP experience, when DCF is complemented by fragmentation. Figure 45 (a) shows that large fragments of 1024 KB improve/reduce the packet delay variation, whereas smaller fragments of 256 KB increase the packet delay variation. Figure 45(b) highlights the fact that fragmentation does not affect the end to end delay. Interestingly, there is no effect on the MOS although the packet delay variation changes when fragmented is introduced to a DCF supported system.



(a)Voice Packet Delay Variation



(b) Voice packet End to end delay

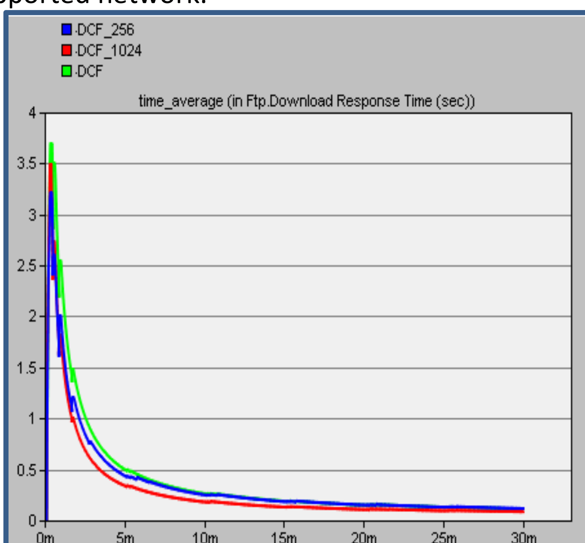


(c)MOS

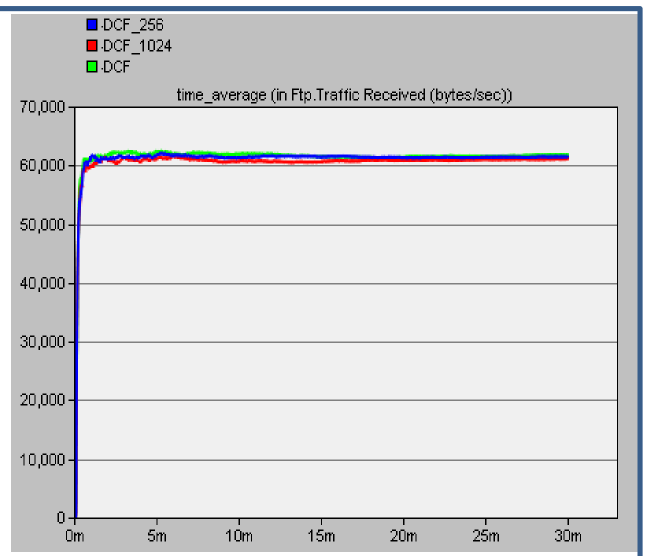
Figure 45: The effect of fragmentation on DCF support for VoIP traffic.

ii. FTP Performance

The Figure 46 below shows the results of introducing fragmentation to a DCF supported network. Figure 46(a) shows that fragmentation does not affect the FTP downloading Response time. Figure 46(b) also shows that fragmentation does not improve nor degrade FTP throughput when it is introduced in a DCF supported network.



(a)Download Response time.



(b) FTP Traffic throughput.

Figure 46: The effect of fragmentation on DCF for supporting FTP traffic.

6.1.3.4. Analysis

VoIP employs the play-out buffer to minimise the effects of jitter/delay variation, this may explain why in spite of the fact that packet delay variation changes when fragmentation is employed alongside DCF, but to significant impact is observed on the MOS.

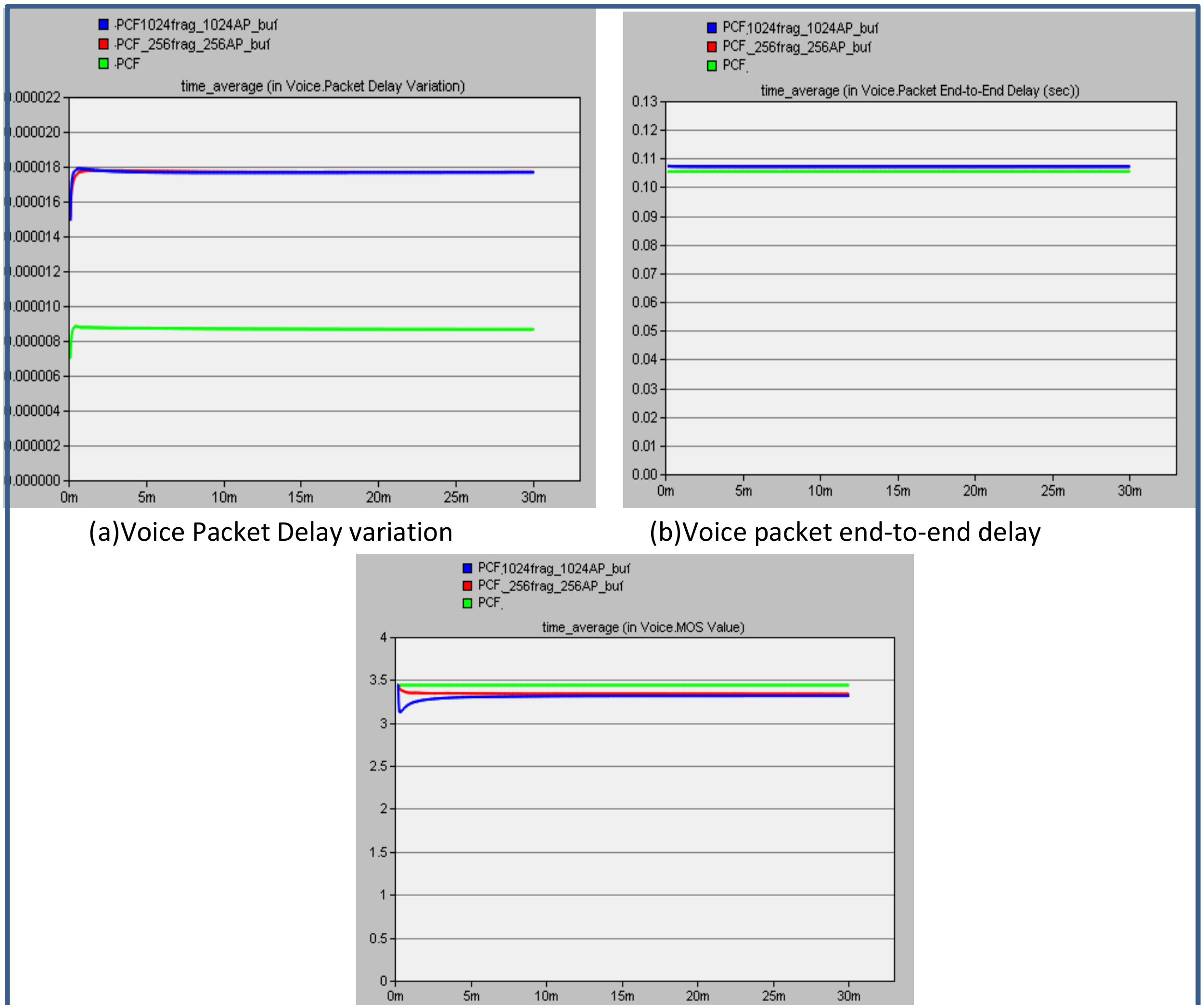
6.1.4. Access Point Buffer Size Variation

The researchers in article [89] concluded that the Access Point by virtue of the fact that it is the relaying station, in MAC that gives all stations an equal opportunity access to the medium; it can potentially become the bottleneck. A simulated study in article [92] found that the buffer length of the Access Point plays an important role in bandwidth allocation. This section presents the results of an investigation into the effect of the Access Point buffer size on the performance of the three coordination schemes in a network where there is an equal number of VoIP and FTP stations. Earlier investigations used a default buffer length of 64KB. In this section the buffer size is increased to 256 KB and further to 1024 KB, while fragmentation is still in service and the performance of each of the coordination schemes, with respect to carrying VoIP and FTP traffic is investigated.

6.1.4.1. PCF

i. VoIP Performance

The Figure 47(a)-(c) below shows the performance of PCF when it is required to manage medium access with Access Point buffer sizes of 256 KB and 1024 KB. Figure 47(a) shows that the increasing the Access Point buffer size from 64 KB to either 256 KB or 1024 KB degrades the packet delay variation, i.e. the packet delay variation increases. Figure 47(b) shows that increasing the Access point buffer also increases the voice end to end delay. Lastly, as expected the resulting MOS for an increased Access Point buffer is relatively poorer. These results suggest that incrementing the buffer size of the Access point brings no additional value to voice services when PCF is employed.



(a)Voice Packet Delay variation

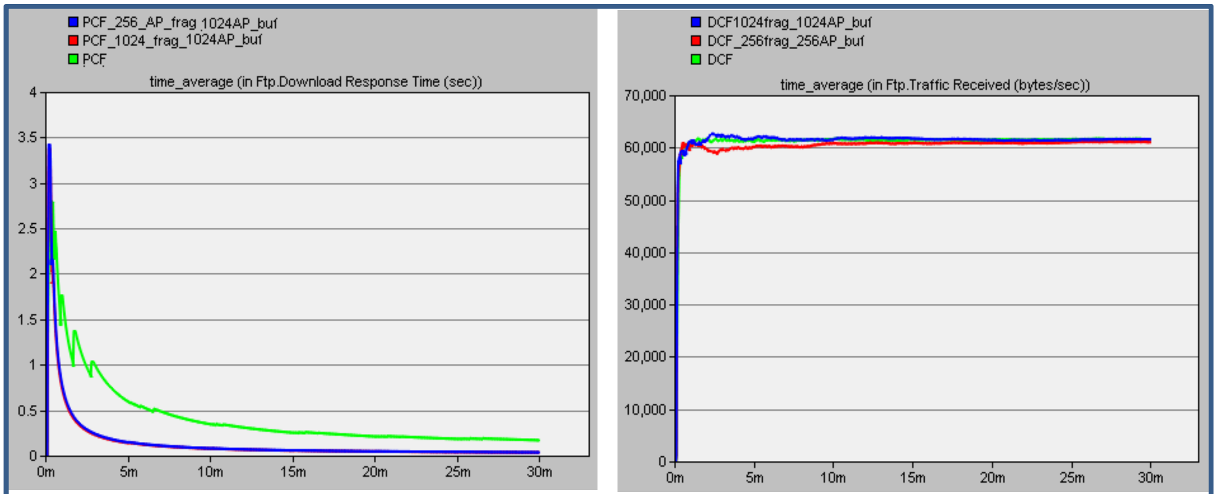
(b)Voice packet end-to-end delay

(c) MOS

Figure 47: Effect of buffer size variation on PCF performance for VoIP

ii. FTP Performance

The Figure 48 below shows the effect of increasing the Access Point buffer size on the performance of PCF with respect to carrying FTP traffic. Figure 48(a) shows that by increasing the Access Point buffer length to 256 KB the FTP download response time can be improved. The Figure 48(a) also shows that increasing the size of the Access Point buffer beyond the 256KB to 1024 KB does not bring additional advantage, but merely takes away valuable memory resources. Figure 48 (b) shows the overall FTP throughput when the buffer size is increased and the diagram demonstrates that in the long-term, increasing the buffer size does not really improve the throughput of FTP data traffic.



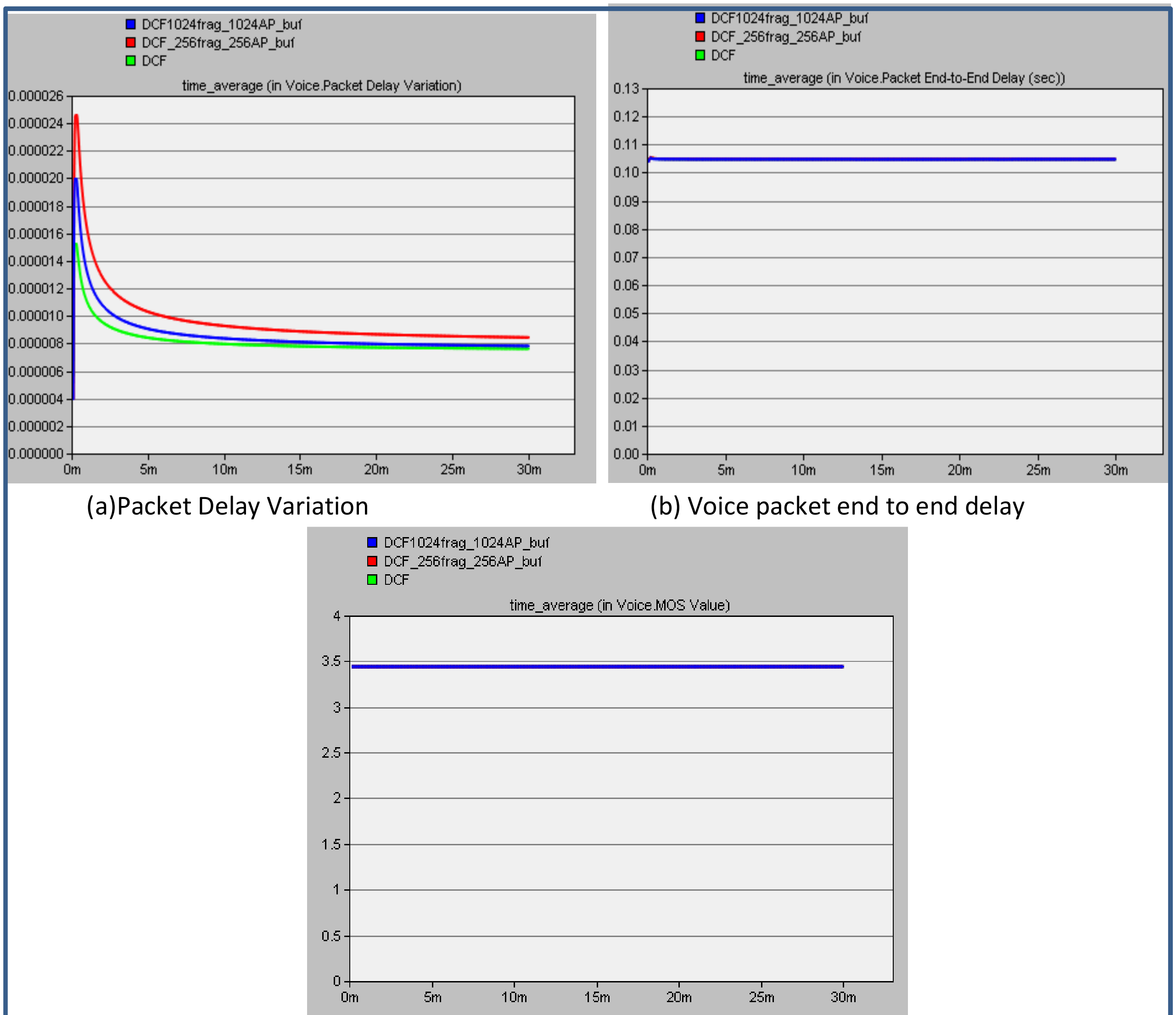
(a)FTP Download Response time (b)FTP Received traffic Throughput

Figure 48: Effect of buffer size variation on PCF performance for FTP

6.1.4.2. DCF

i. VoIP Performance

The Figure 49 shows the effect of lengthening the Access Point buffer size on DCF's ability to carry VoIP traffic. Figure 49(a) shows that increasing buffer length while fragmentation is in service does not materially improve the packet delay variation. However, the combination of 256 KB Access Point buffer and 256 KB maximum fragment deteriorates the packet delay variation. Figure 49(b) and (c) show that no additional end to end delay and MOS value accrues from increasing the buffer length. Therefore, increasing the buffer length when DCF is in services also uses critical memory resources without necessarily bringing additional performance improvement. What is also notable is that the deterioration in packet delay variation as a result of the 256 KB buffer did not materially affect the overall MOS.



(a)Packet Delay Variation

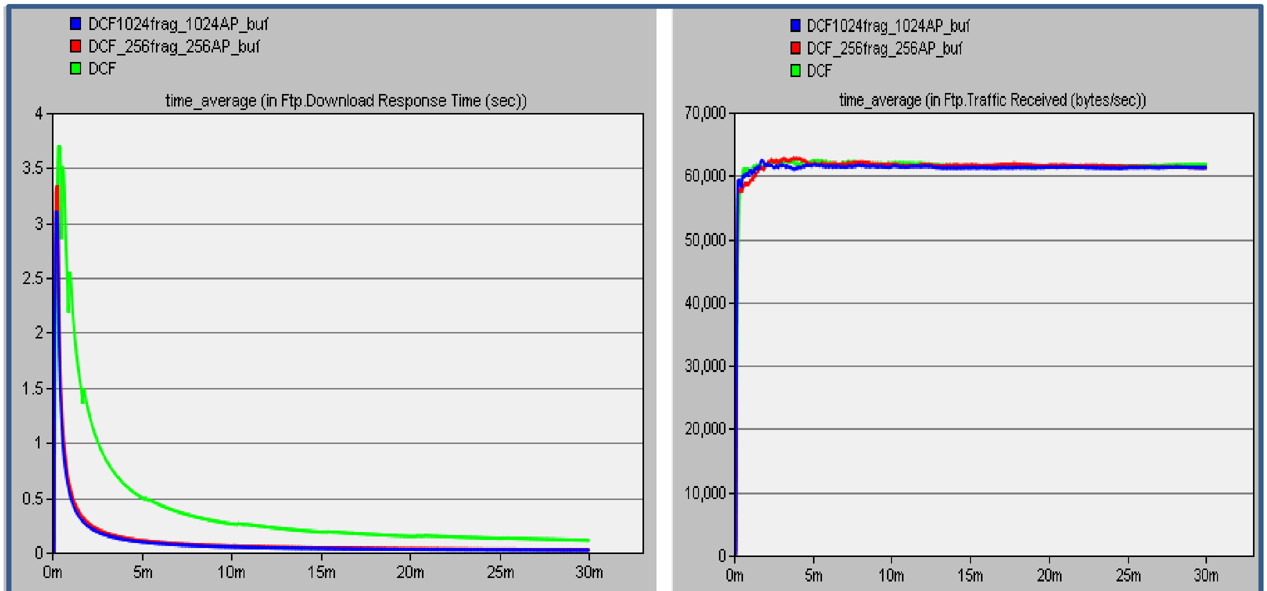
(b) Voice packet end to end delay

(c) MOS

Figure 49: Effect of buffer size variation on DCF performance for VoIP

ii. FTP Performance

The results of increasing the size of the Access Point buffer on DCF's ability to carry FTP traffic are presented below. Similar to PCF, increasing the buffer length improves the FTP response time. However, it has no effect on the overall FTP throughput as highlighted in Figure 50. Therefore, when DCF is employed, increasing the Access Point buffer length does not bring any additional material benefit for FTP users.



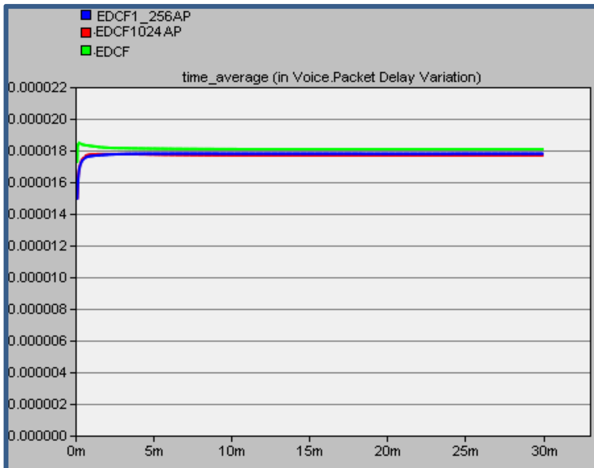
(a) Download Response Time (b) FTP traffic throughput

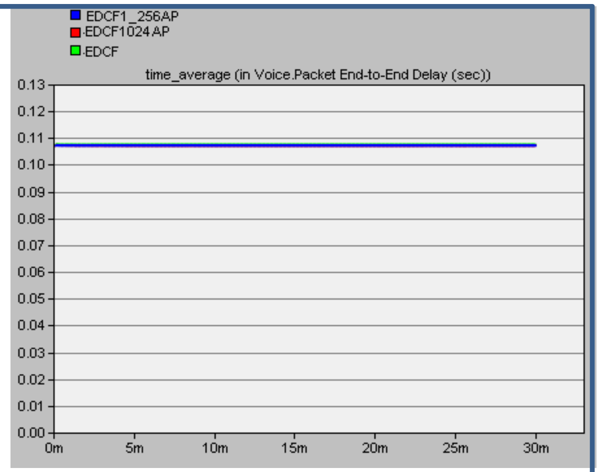
Figure 50: Effect of increasing Access Point buffer on DCF ability to carry FTP traffic.

6.1.4.3. EDCF

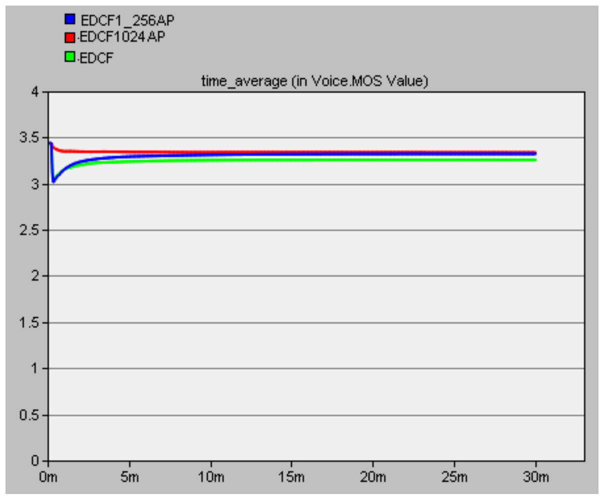
i. VoIP Performance

When the Access Point's buffer length is increased from 64KB to 256KB and on to 1024 KB, the voice packet delay variation is improved, albeit by a very small margin see Figure 51(a), the blue and red curves are slightly below the green curve. The AP buffer does not improve the end to end delay at all. The MOS is also slightly improved by lengthening the buffer of the Access Point. However, it does not seem as though increasing the buffer length beyond 1024 KB brings any additional improvements to the performance of EDCF.





(a)Packet Delay Variation (b) Voice packet end to end delay

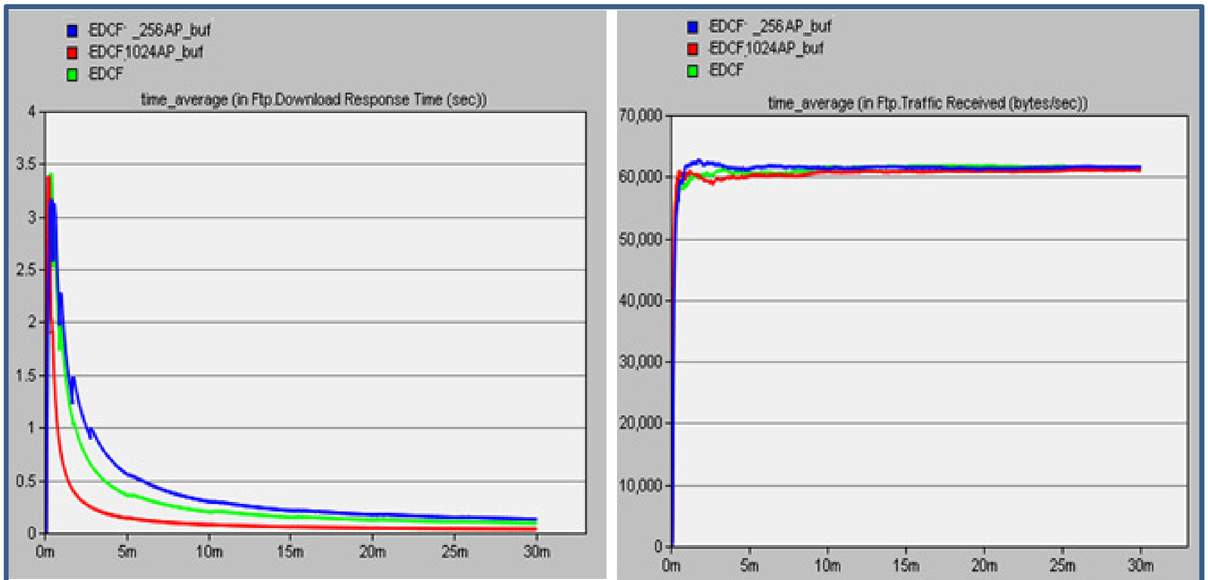


(c) MOS

Figure 51: Effect of increasing Access Point buffer on EDCF ability to carry VoIP traffic

ii. FTP Performance

In section 6.1.1 the performance of EDCF was compared to that of PCF and DCF when the Access Point buffer length was set to 64 KB and it was found that EDCF performs the worst with respect to FTP traffic. This section then asked the question, can the performance of EDCF improve if the Access Point buffer is lengthened? The results are presented in Figure 52(a) and (b). In the Figure 52(a) it is clear that the 1024 KB buffer length improves the FTP download response time by a very small amount, whereas the 256 KB buffer length degrades the performance albeit, only slightly. There is also no improvement in performance of with respect to FTP traffic throughput. Therefore, the performance of EDCF cannot be improved by increasing the Access Point buffer length.



(a). FTP Download response Time (b) FTP traffic throughput

Figure 52: FTP performance of EDCF with increased AP buffer length

6.2. 45 FTP Stations and 15 VoIP Stations.

The previous section 6.1 presented an investigation into the performance of PCF, EDCF and PCF with respect to carrying VoIP and FTP data when there is an equal split between VoIP and FTP stations. The investigations were further extended by allowing only VoIP station to participate in the CFP and increasing the duration of the contention free period. The size of the frames was identified as a potential efficiency issue in lossy wireless networks. Therefore the effect of fragmenting frames was investigated for each one of the stations. The Access Point was also identified as a potential bottleneck and the effect of increasing the buffer length of the Access Point in order to improve performance was investigated for each of the coordination schemes.

This section of the report re-considers the above investigation when the network is dominated by stations that are performing FTP services. The purpose of this investigation is to understand, to what effect do FTP connection cannibalise network resources when there is a fewer number of VoIP stations and how do the three coordination schemes compare with respect to managing the medium access fairly to ensure that both types of users are fairly services. For an operator the voice service is more important because of its real-time requirements. Therefore it is important to understand how FTP stations affect the overall VoIP quality of experience when there are fewer VoIP stations.

6.2.1. PCF, EDCF and DCF with no enhancements.

6.2.1.1. VoIP Performance

In section 6.1.1 it was found that in a network where there are an equal number of Voice and FTP traffic stations, the best VoIP performance is achieved by DCF. It was also found that PCF does not improve the performance, while EDCF gives a degraded performance.

The Figure 53 below outlines the performance when the FTP stations dominate the network. Figure 53(a) presents the average packet end to end delay, but this presentation does not highlight the difference between the three schemes. A frequency representation as in Figure 53(b) shows that DCF still gives the best end to end delay. Figure 53(c) indicates that DCF still gives the best packet delay variation of all the schemes. However, Figure 53(d) shows that the superior packet delay variation performance by DCF does not necessarily translate into a superior overall experience as highlighted by the MOS. The minimal effect of the packet delay variation can be attributed to the use of playback buffer in VoIP systems. More interestingly, the MOS is equal across all the schemes; unlike in the first case where there was an equal number of VoIP and FTP stations and EDCF gave the worst MOS relative to both DCF and PCF. This means that when there is a fewer number of VoIP stations relative to FTP stations, EDCF gives the best performance improvement of all the three coordination schemes.

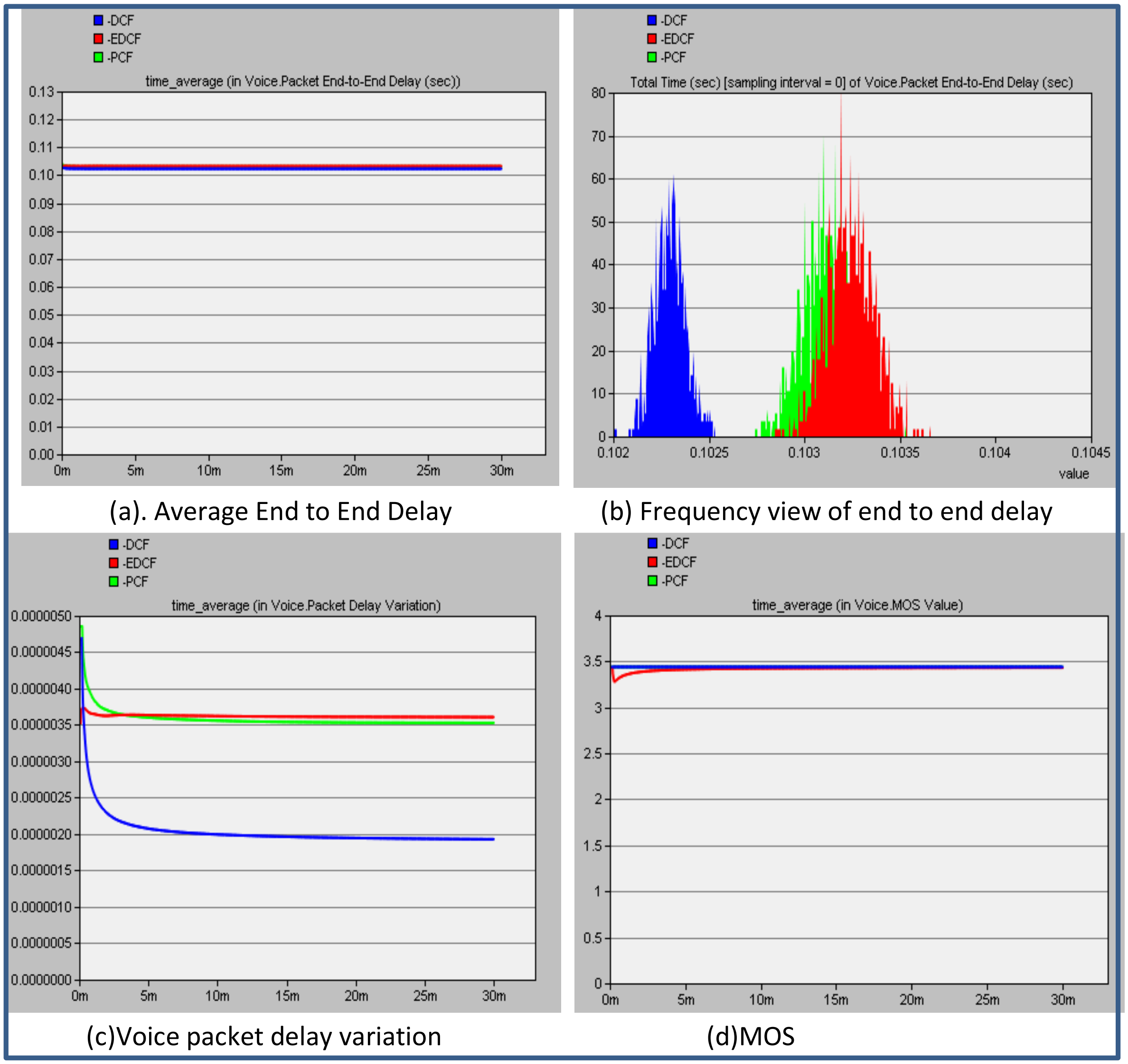


Figure 53: Performance of PCF, DCF and EDCF for VoIP.

6.2.1.2. FTP Performance

In section 6.1.1.1, it was found that PCF gives the worst FTP download response time, but this was by a fairly small margin. With regards to FTP traffic throughput there was not a material difference between the three schemes, all three gave an average throughput of 60Kbps. Figure 54(a) indicates a materially poor

FTP response time performance by PCF relative to both DCF and EDCF. Therefore, PCF performance is degraded relative to both DCF and EDCF when the BSS is dominated by FTP stations. This means that PCF is unsuitable for carrying FTP traffic when they dominate the network. This is because the PCF's polling queue for FTP stations is increased and as such this reduces the response time.

With regards to FTP throughput all three coordination schemes show a materially improved throughput from 60Kbps to 90 Kbps. There is still no major difference in the throughput across the three schemes. The improved FTP throughput may be attributed to the fact that there are a lesser number of VoIP stations competing for the limited access to the medium.

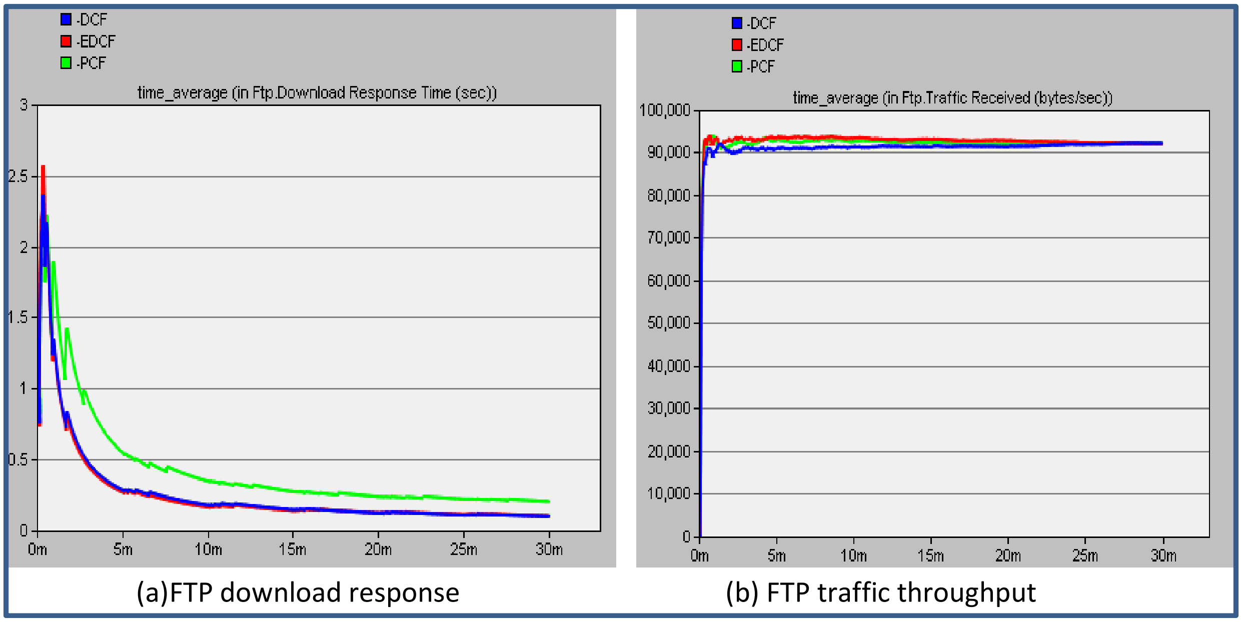


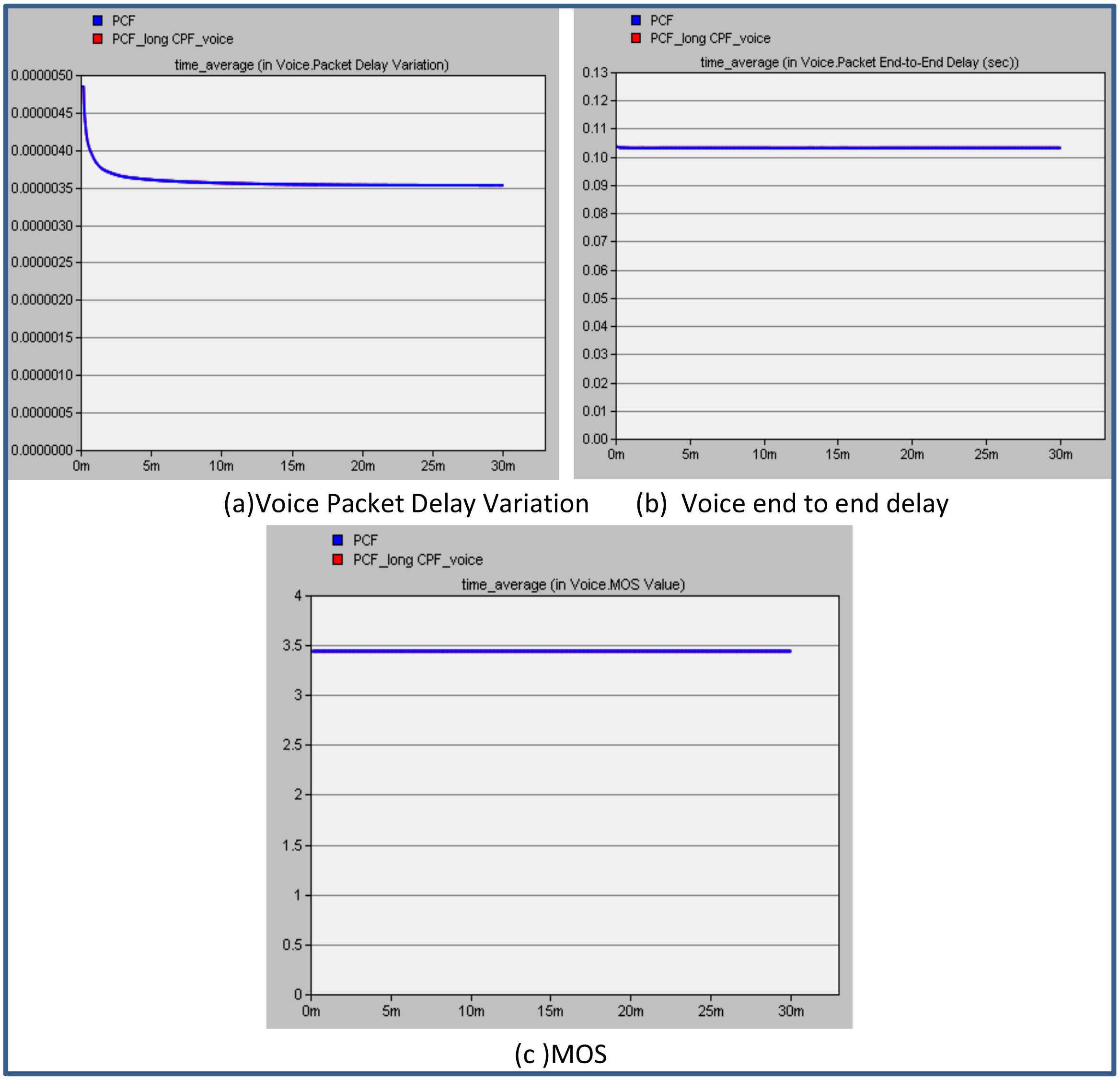
Figure 54: Performance of PCF, DCF and EDCF for carrying FTP traffic.

6.2.2. Effect of increasing CFP and allowing only VoIP stations to transmit during CFP.

When there was an equal number of VoIP and FTP stations in section 6.1.2, lengthening the Contention Free period(CFP) degraded the performance of PCF by degrading quality of experience MOS, end to end packet delay variation and end to end packet delay for voice while the FTP experience download response time and the FTP traffic throughput. In this section PCF is re-evaluated when there is a fewer number of VoIP stations.

6.2.2.1. VoIP Performance

Figure 55 (a)-(c) indicates that increasing the CFP in a BSS that is dominated by FTP stations brings no additional value to VoIP nor does it degrade the performance of the network, relative to all stations participating in a shorter 0.01 CPF. When there were an equal number of stations with a lengthened CFP and only VoIP station participated, the lengthened CFP degraded MOS from about 3.49 to 3.25, Figure 39. However, Figure 55(c) shows that the MOS remained at 3.49 as a result of increasing the CFP when there are more FTP stations than VoIP stations i.e. 25/75 BSS. Therefore lengthening PCF when there is a greater number of FTP stations do not improve nor degrade performance, although it does degrade performance when there is an equal number of stations.



(a)Voice Packet Delay Variation (b) Voice end to end delay

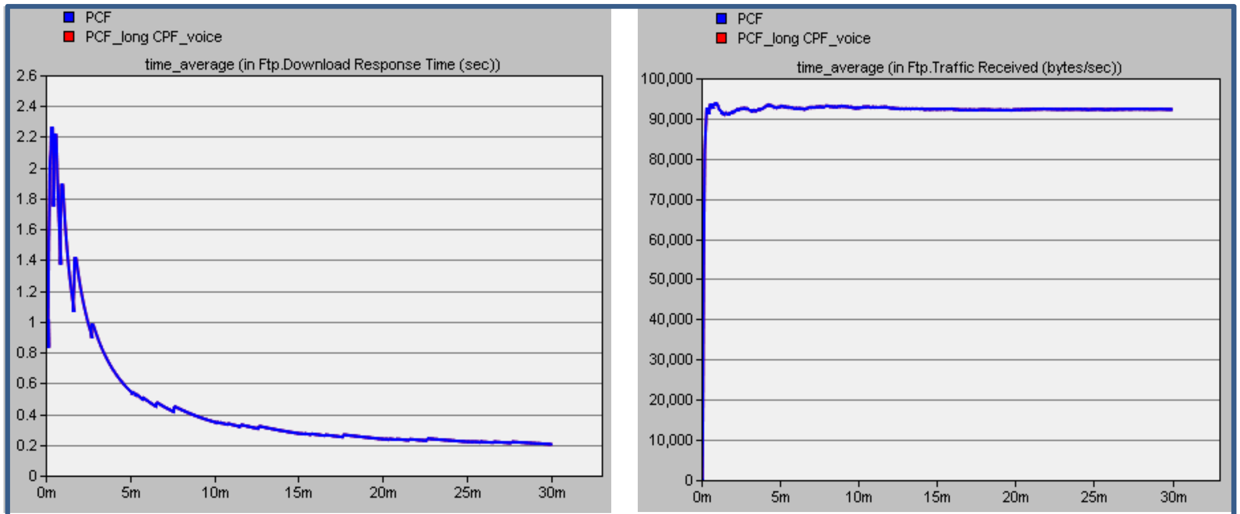
(c)MOS

Figure 55: Performance of PCF with elongated CFP

6.2.2.2. FTP Performance

When there were an equal number of VoIP and FTP stations in 6.1.2.2, lengthening the CFP and allowing only VoIP stations to participate improved the FTP response time and only slightly improved the FTP traffic throughput.

The Figure 56(a) and (b) outlines the performance of PCF with regards to carrying FTP traffic when the VoIP station dominate the BSS and only VoIP stations participate in the CFP. Unlike in an equal station scenario where lengthening the CFP improved the FTP download response time, no improvement is achieved by lengthening the CFP when there is a greater number of FTP stations compared to VoIP stations. However, the FTP traffic throughput improved materially from 60Kbps to 93Kbps when there are a greater number of FTP stations. This is somewhat expected because during the Contention Period FTP stations will routinely win the contest, but because there is a fewer number of VoIP stations, the VoIP performance is not materially affected as seen in 6.2.2.1.



(a) FTP Download response time (b) FTP traffic throughput

Figure 56: Performance of PCF with lengthened CFP for VoIP.

6.2.3. Fragmentation for voice stations with Maximum 256 KB and 1024KB Fragments.

In section 6.1.3 the results of introducing layer-2 fragmentation with maximum fragment lengths of 256KB and 1024KB was presented for a case wherein there is an equal number of VoIP stations and FTP stations. The effect of fragmentation was analysed for the three MAC schemes, PCF, DCF and EDCF. This section repeats the above investigation when there are a greater number of FTP stations compared to VoIP stations i.e. 15 VoIP and 45 FTP.

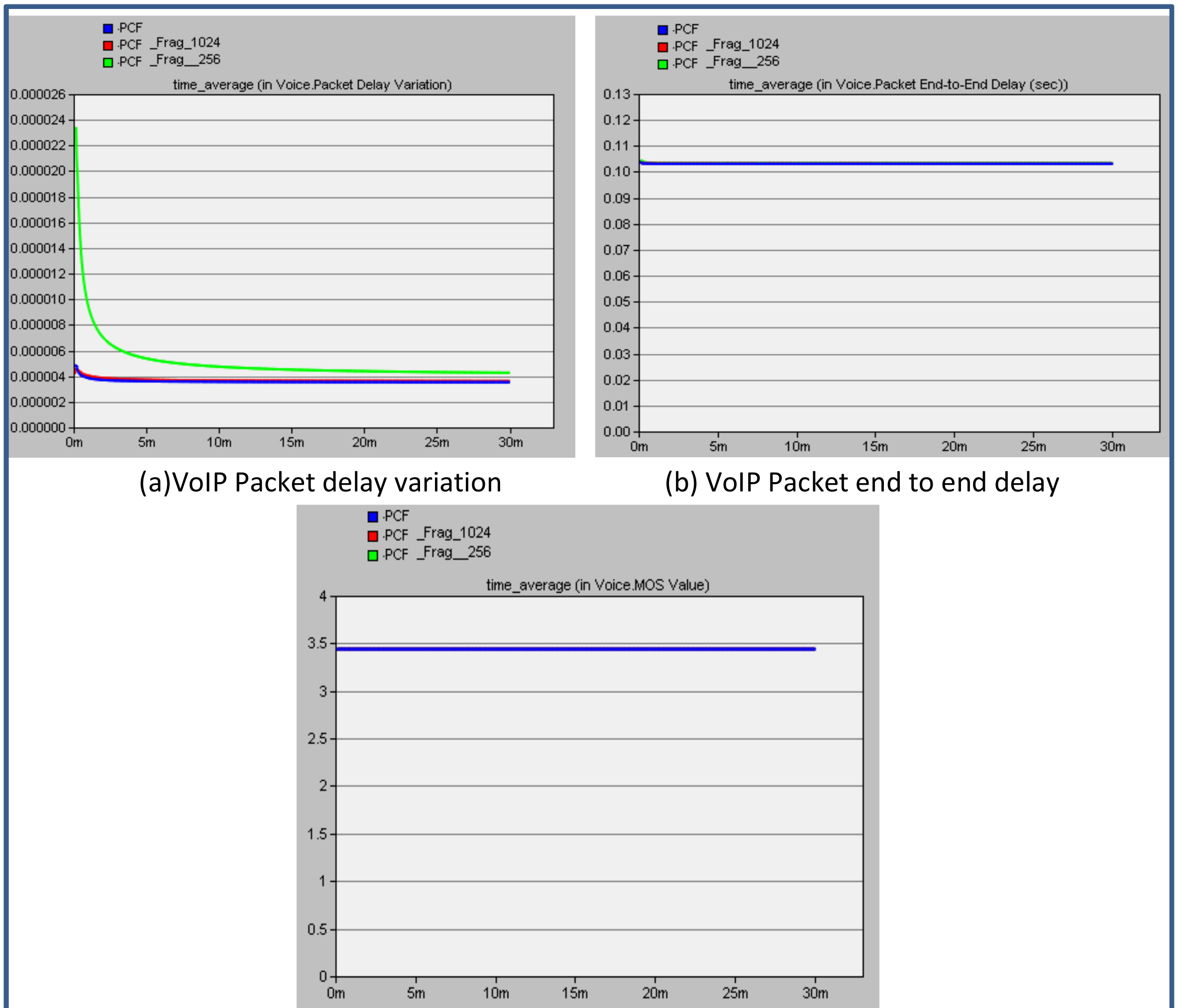
6.2.3.1. PCF

i. VoIP Performance

PCF with fragmentation improved the VoIP end to end variation for both maximum fragment lengths when there were an equal number of stations i(i). PCF with fragmentation further improved the MOS when there was an equal number of stations and a maximum fragment length of 1024 KB i(i). There was no observed change in the end to end delay variation i(i).

Figure 57(a)-(c) presents the VoIP results of using fragmentation to supplement PCF when there are a greater number of FTP stations compared to VoIP stations. In Figure 57 (a) it can be observed that when FTP stations dominate the BSS, fragmentation with maximum fragment length of 256 KB degrades packet delay variation, whereas the maximum fragment length of 1024 KB does not improve performance above PCF with no fragmentation. There is no notable change in VoIP packet end to end delay and MOS value.

Therefore, when FTP stations dominate the BSS fragmentation increases the processing demands on the stations but it does not bring any material improvement in VoIP performance.



(a)VoIP Packet delay variation

(b) VoIP Packet end to end delay

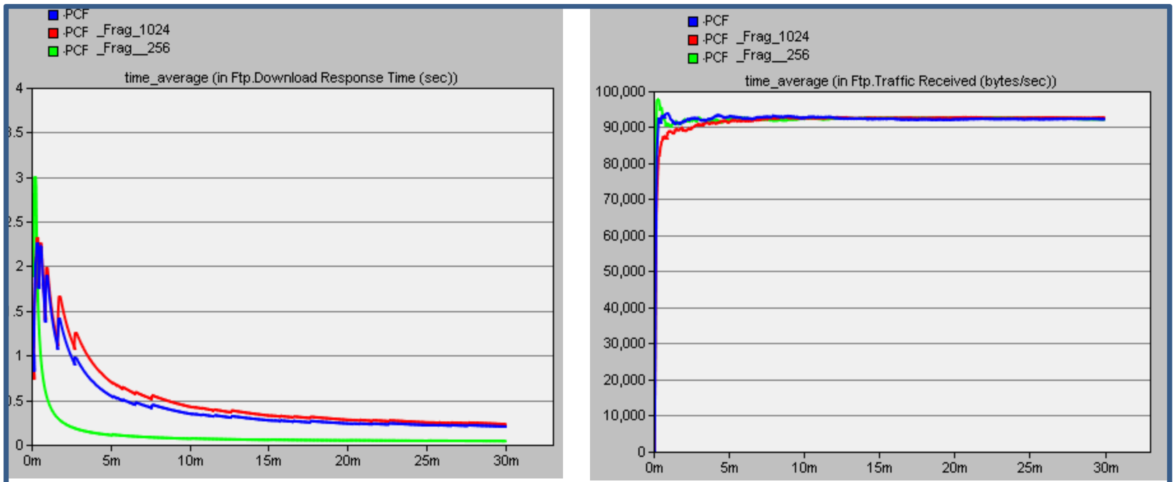
(c) MOS

Figure 57: VOIP performance with fragmentation

ii. FTP Performance

The effect of introducing fragmentation to PCF on FTP traffic performance for equal number of stations was presented in 6.1.3.1(i). In that section it was observed that fragmentation of any size does not bring any additional performance improvement and as such it introduces unnecessary processing demands on the stations.

Figure 58(a) and (b) presents similar results for a BSS that is dominated by FTP stations. The results show that the 256 KB maximum fragment length improves the FTP download response time. However, it does not improve the FTP traffic throughput over PCF without fragmentation.



(a). FTP Download response time (b) FTP traffic throughput

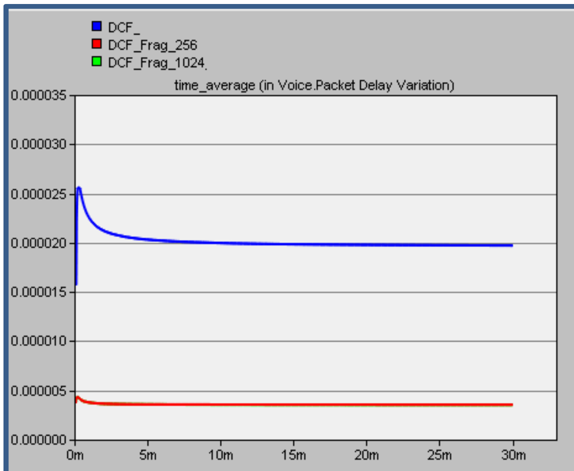
Figure 58: FTP Performance with Fragmentation

6.2.3.2. DCF

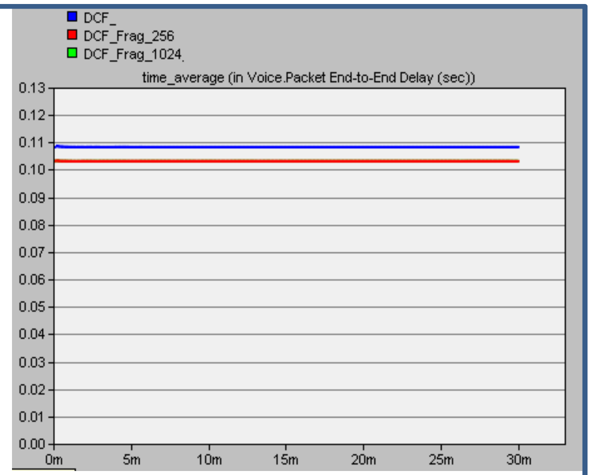
i. VoIP Performance

In 6.1.3.3, where there was an equal number of VoIP and FTP station, large frame fragmentation of 1024KB seemed to bring about a material improvement in voice packet delay variation. However, there was no material improvement in either the MOS or the end to end delay.

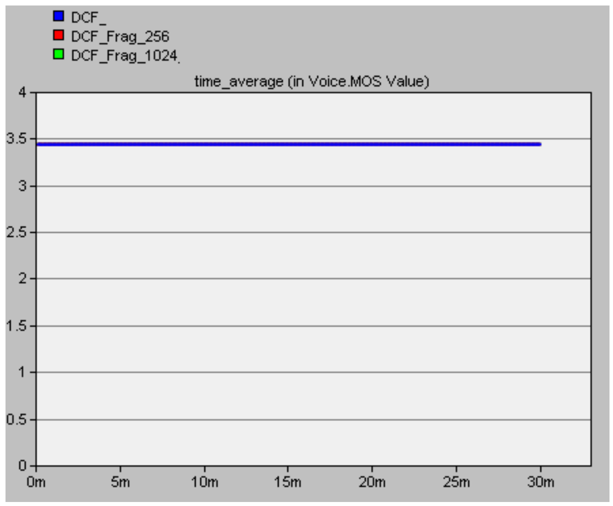
In this section Figure 59(a) –(c) shows the results of fragmentation effect on DCF when FTP stations dominate the network. Figure 59(a) shows clearly that fragmentation has improved the end to end delay variation. It is also clear that increasing the size of fragments beyond 256 KB does not bring additional improvements to the end to end delay variation. Therefore using fragmentation in this scenario reduces the need for a jitter playback buffer. Figure 59(b) shows that by introducing fragmentation to DCF the average end to end delay has improved. However, Figure 59(c) shows no improvement in the overall VoIP experience as a result of fragmentation. This is rather strange given that the end to end delay variation has improved and the end to end delay has improved. This suggests that either fragmentation introduced additional packet-loss or that for the chosen codec G.711, DCF cannot achieve a better MOS value. Fragmentation by its nature reduces packet-loss because large frames which may overload the buffer thus leading to excessive packet drops are reduced in size. Hence it is unlikely that packet loss causes the MOS to remain at the value that was obtained without fragmentation. Unfortunately, the free OPNET version does have an option to assess packet-loss.



(a) Packet delay variation



(b) Packet end to end delay

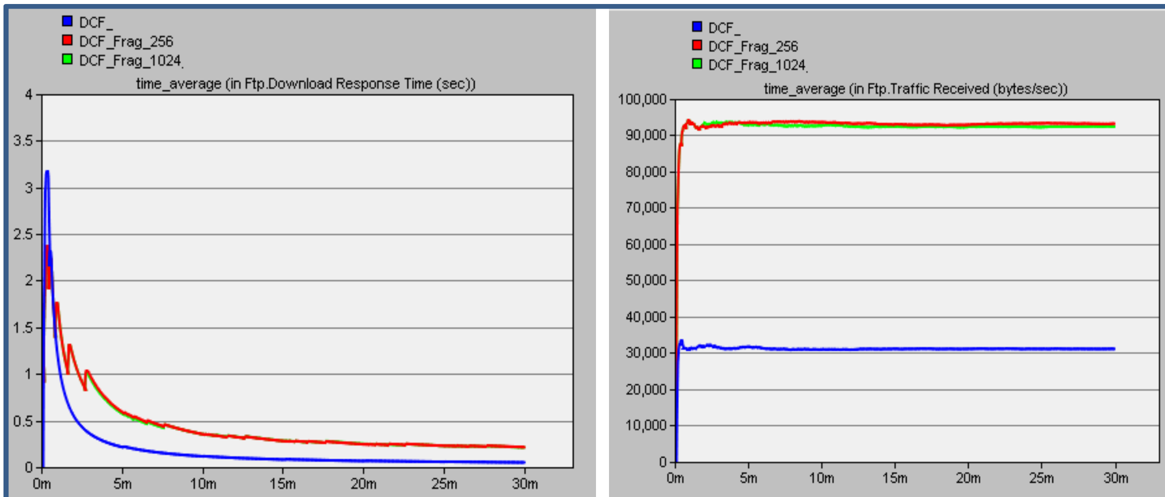


(c) MOS

Figure 59: VoIP performance of DCF for an FTP dominated BSS.

ii. FTP Performance

In section 6.1.3.3 where the number of FTP and VoIP stations was equal, it was found that introducing fragmentation did not affect the performance of FTP stations. Figure 60(a) shows that employing fragmentation in a DCF managed BSS wherein the FTP station dominates the VoIP stations degrades the FTP download response time. The more interesting result is observed in Figure 60(b) which shows that by introducing fragmentation, the overall FTP throughput was improved three folds from about 30 Kbps to 93Kbps. Moreover, increasing the size of fragments beyond 256 KB does not bring any additional performance improvement. When a larger frame is transmitted and an error is experienced due to poor channel conditions, thus requiring a retransmission, this would mean retransmission of the same large frame again. However, if a large frame is broken up then only a small fragment needs to be retransmitted, hence the improved FTP performance.



(a) FTP download Response (b) FTP traffic throughput

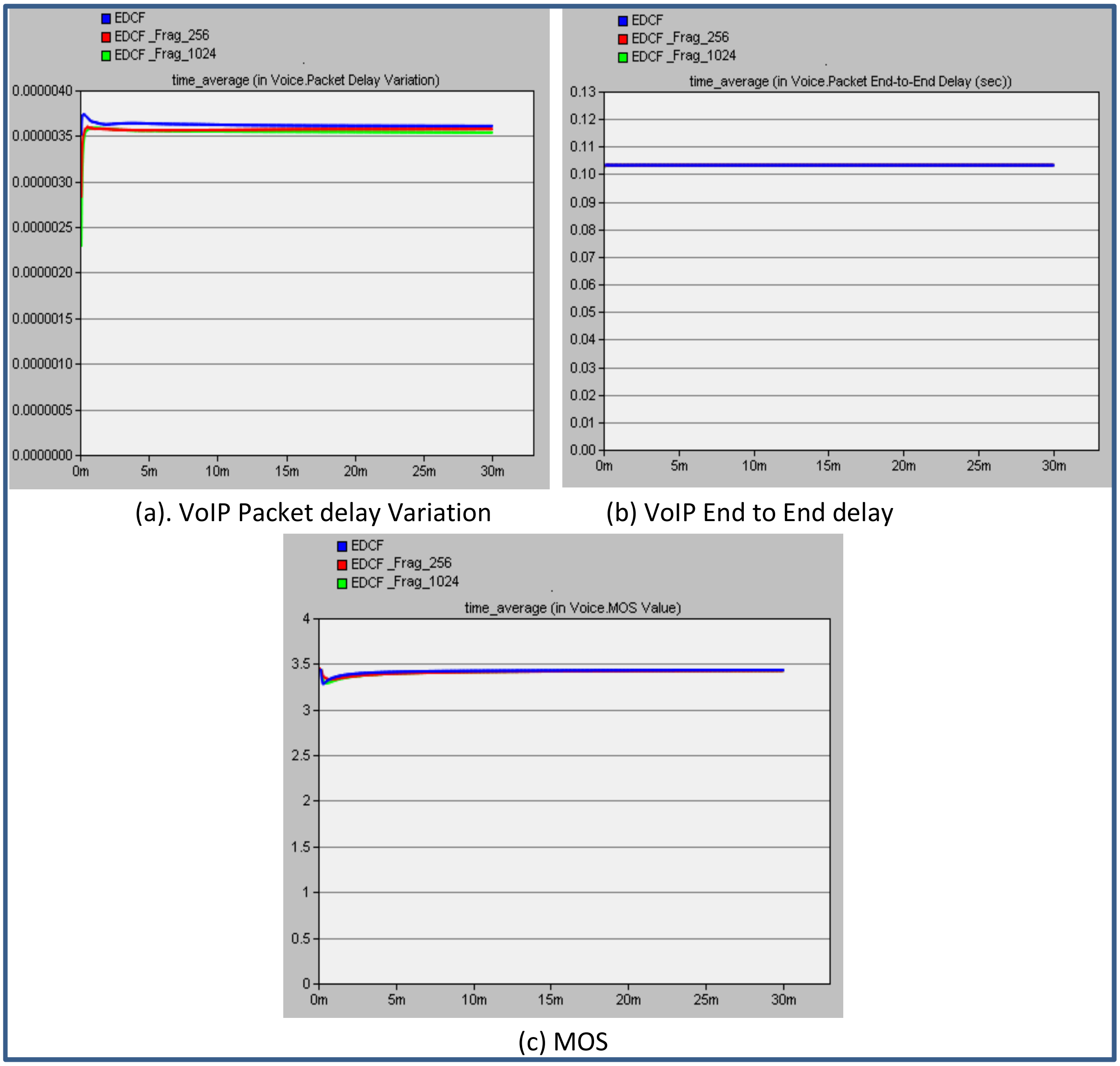
Figure 60: FTP performance of DCF for an FTP dominated BSS

6.2.3.3. EDCF

i. VoIP Performance

In section 6.1.3.2.(i), it was found that introducing fragmentation to EDCF when there is an equal number of VoIP and FTP stations improved the packet end to end delay, the packet delay variation and the MOS. However, the improvements could only be seen if the frequency plot was used. This section presents the VoIP performance when there are an equal number of stations.

In Figure 61(a) it is observed that the packet delay variation was degraded as a result of introducing fragmentation for both maximum fragment lengths of 256 KB and 1024 KB. The Figure 61(b) and (c) show that there is no notable improvement in packet end to end delay and MOS. Therefore, the performance of EDCF is not really improved by introducing any level of fragmentation. This merely presents additional processing for the Access Point and the stations.



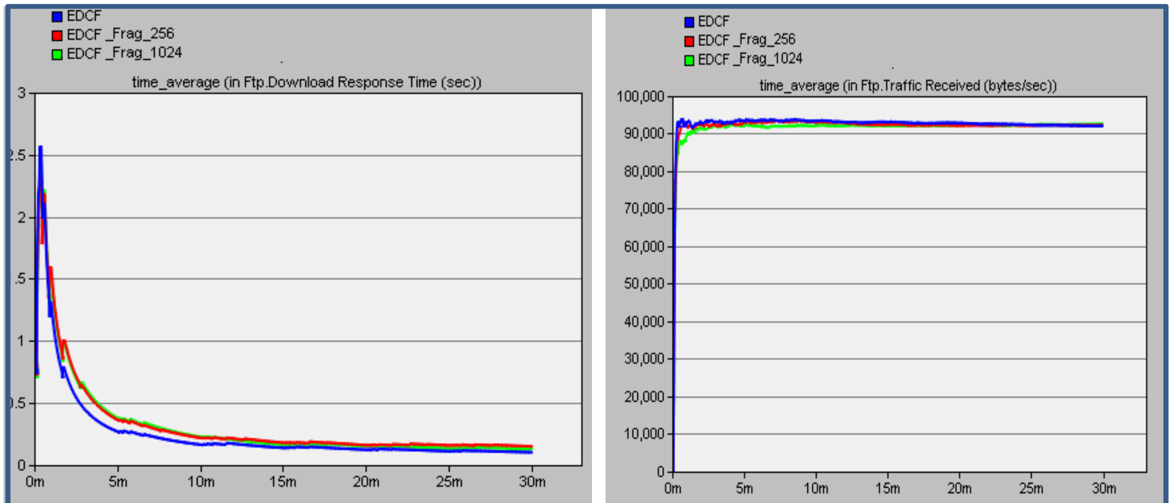
(a). VoIP Packet delay Variation (b) VoIP End to End delay

(c) MOS

Figure 61: EDCF VoIP performance with fragmentation.

ii. FTP Performance

According to the Figure 62(a) below there is a minute improvement in FTP download response performance as a result of introducing fragmentation to an EDCF managed BSS. However, Figure 62(a) shows no improvement in FTP traffic throughput as a result of fragmentation.



(a). FTP Download Response Time (b) FTP Traffic throughput

Figure 62: EDCF FTP performance with fragmentation

6.2.4. Access Point Buffer Size Variation

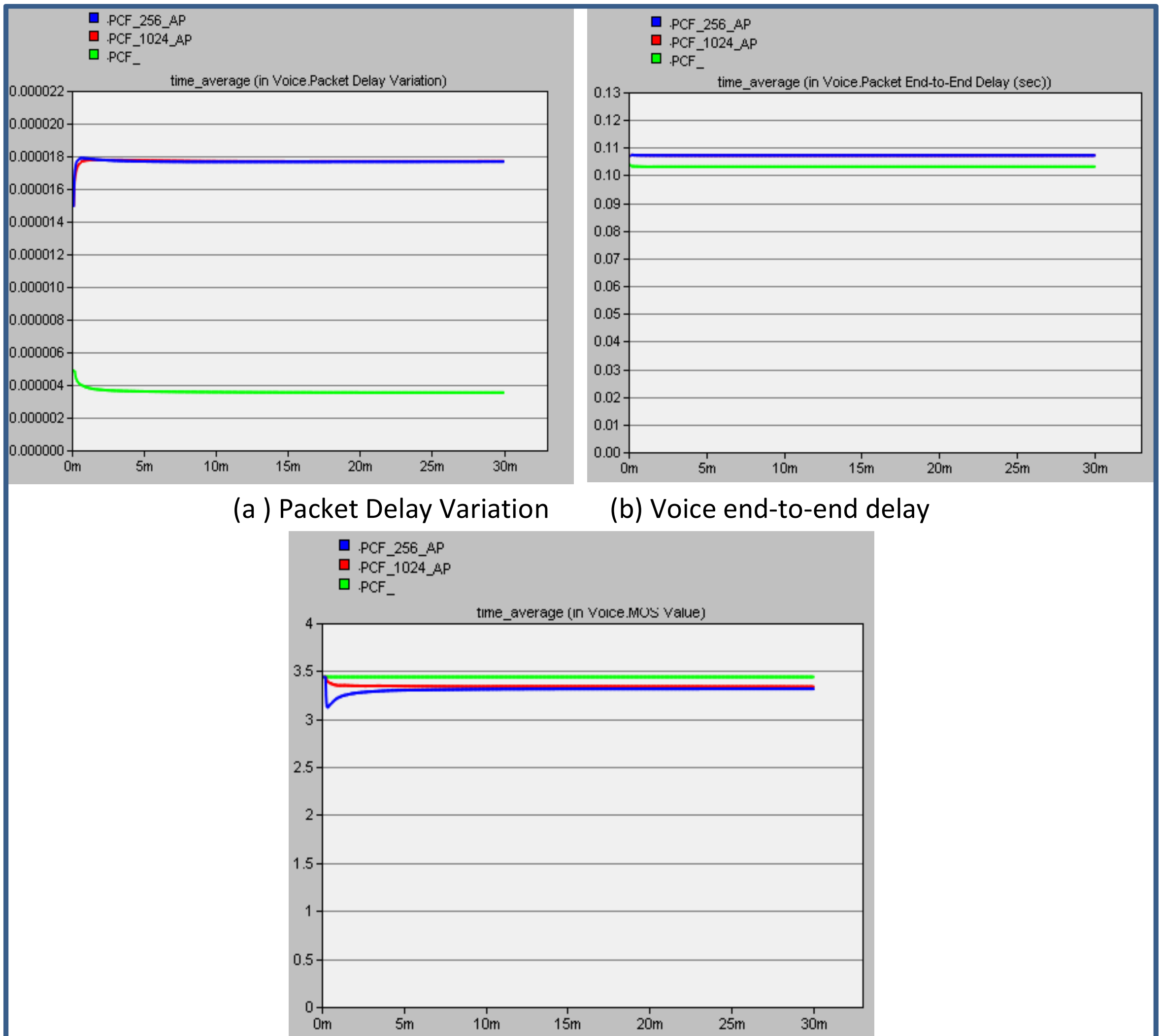
In section 6.1.4 the effect of increasing the buffer size of the Access Point on the ability of PCF, DCF and EDCF to carry FTP and VoIP traffic over the same Access Point was evaluated under the condition that the number of VoIP station equal the number of FTP stations. The effect of increasing the buffer size of the Access point was evaluated due to the fact that the Access Point has to relay frames from multiple stations and there it buffer may be getting filled and as such dropping packets.

In this section the effect of increasing the buffer size is again investigated for a situation wherein the FTP stations far outnumber the VoIP stations by 45/15.

6.2.4.1. PCF

i. VoIP Performance

Figure 63 below outlines the effect of increasing the buffer size on the ability of PCF to carry VoIP traffic. Figure 63(a) shows that increasing the buffer length to 256KB and 1024 KB respectively degrades the packet delay variation. Furthermore, Figure 63(b) also shows that increasing the buffer size degrades the end to end delay. Therefore as expected the MOS in Figure 63(c) is also degraded. The same observations were noted in 6.1.4.1. Therefore, the buffer length reduces the on the ability of PCF to carry voice when the number of FTP stations far exceeds the number of VoIP stations.



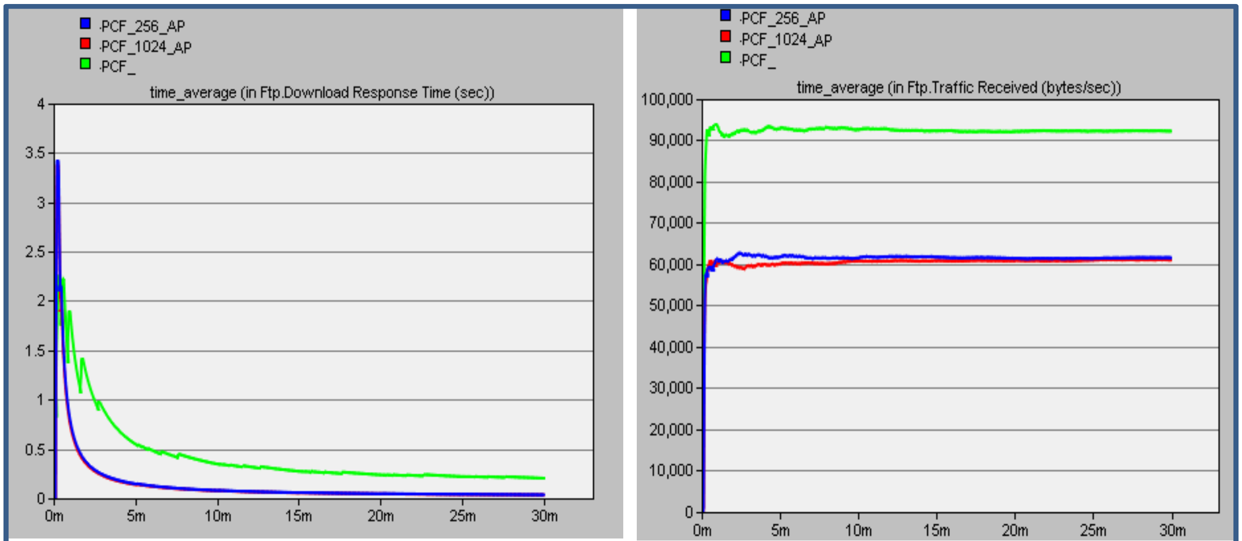
(a) Packet Delay Variation (b) Voice end-to-end delay

(c) MOS

Figure 63: VoIP performance of PCF for an FTP dominated BSS

ii. FTP Performance

Increasing the buffer size has clearly improved the FTP download response time; however, it is to be noted that the length of the buffer size beyond 256 KB does not bring additional value to FTP response time. On the other hand, the FTP throughput was severely curtailed by increasing the buffer size from 64 KB to 256 KB and on to 1024 KB. The FTP traffic throughput dropped from above 90 Kbps to about 60Kbps.



(a) FTP Download response time (b) FTP traffic throughput

Figure 64: FTP performance of PCF for an FTP dominated BSS

6.2.4.2. DCF

i. VoIP Performance

In 6.1.4.2 when the BSS was loaded with 30 FTP and 30 VoIP, it was found that the 256 KB buffer length increased the packet delay variation, but did not affect the MOS. The results of increasing the length of the Access Point buffer, from 64Kbps to 256 Kbps and onto 1024 Kbps, where there is a greater number of FTP stations (45) are indicated in Figure 65. In the Figure 65(a) shows that unlike in an equally loaded network the voice delays variation deteriorates as the buffer size increases, albeit it remains within the acceptable limit of maximum 2%. The Figure 65(b) suggests that there is no change in the end to end delay as a result of increasing the buffer, but the figure remains within acceptable limit of less than 150ms. An alternative frequency plot suggests that increasing the buffer length reduces the end to end delay, albeit by a very small margin. Lastly, there is no visible impact on the MOS.

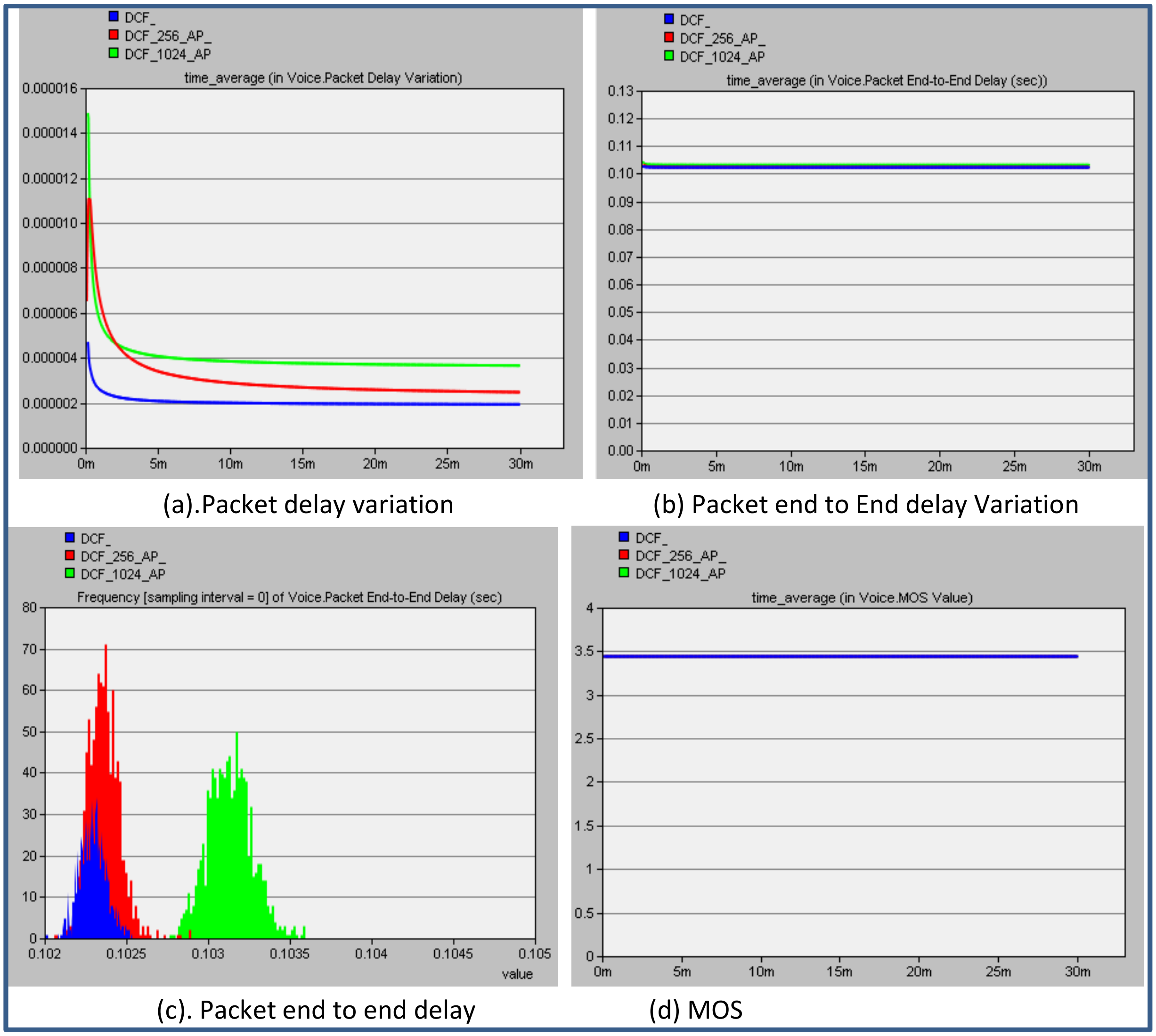


Figure 65: DCF VoIP performance for lengthened Access Point Buffer.

ii. FTP Performance

In section 6.1.4.2(ii) it was found that when there are 30 FTP and 30 VoIP stations there is no effect on throughput, but the FTP response time is improved. The same observation is seen in Figure 66(a) where the FTP response time improved when the Access Point buffer length was increased to 256 KB from 64KB that was used for vanilla DCF. However, a further increase to 1024KB does not bring any additional value. Interestingly the traffic through put for FTP stations has increased from 60Kbps when there were only 30 FTP stations to 90 Kbps. This can be explained by the fact that DCF is a fair contention based mechanism; as such it is likely that if there are more FTP stations, they will win most medium access contests, as such higher FTP traffic will pass through.

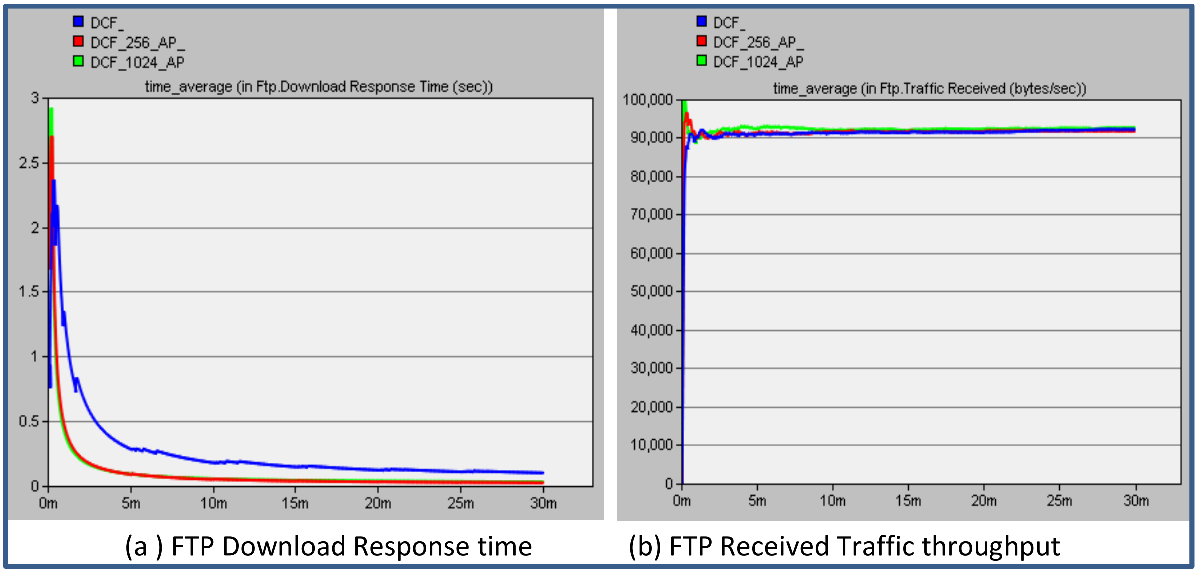


Figure 66: DCF performance with FTP traffic

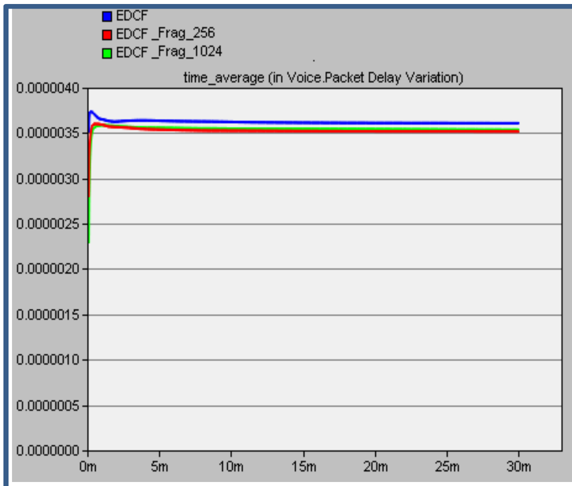
6.2.4.3. EDCF

In section 6.1.4.3 the effect, on both Voice and FTP traffic, of increasing the Access Point buffer for a EDCF managed BSS, when there is an equal number of VoIP and FTP stations, was presented. In that section it was reported that increasing the AP buffer to 256 KB improved the Voice delay variation by a minute value, but a further increase to 1024 KB does not bring any additional improvements.

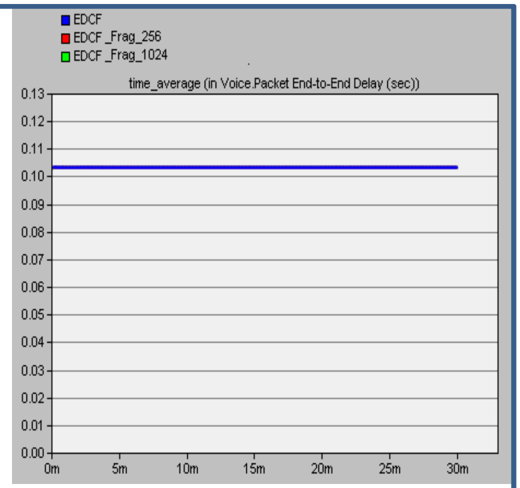
This section presents the results of repeating this evaluation for EDCF with a new conditional that the number of FTP stations exceeds the number of Voice stations, 45 FTP and 15 VoIP.

i. VoIP Performance

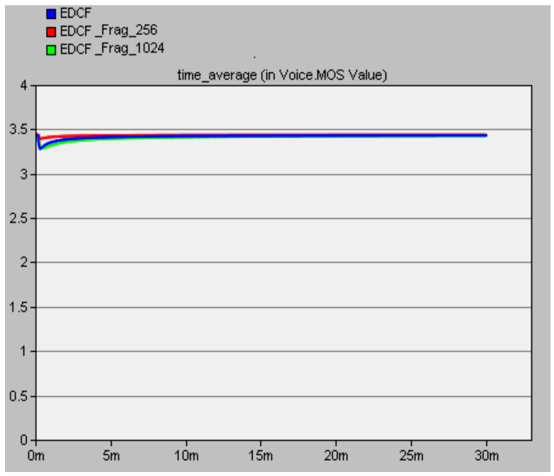
In section 6.1.4.3(i) it was found that only a minute improvement in packet delay variation is observed when the AP buffer length was increased to 256 KB, but beyond this value no improvement could be observed. Furthermore, no visible improvement or degradation in MOS could be observed. In Figure 67(a) a slight degradation is observed when the buffer lengths are increased to 256 KB and on to 1024 KB. Similar to the situation where in there is an equal number of FTP and VoIP stations, no improvement or degradations is observed in MOS and end to end delay variation. Therefore, when there are a greater number of FTP stations in a BSS and EDCF is employed, increasing the buffer size of the Access Point does not bring any additional value to the system; it merely consumes extra memory resources.



(a).VoIP packet delay Variation



(b) VoIP end to end delay Variation

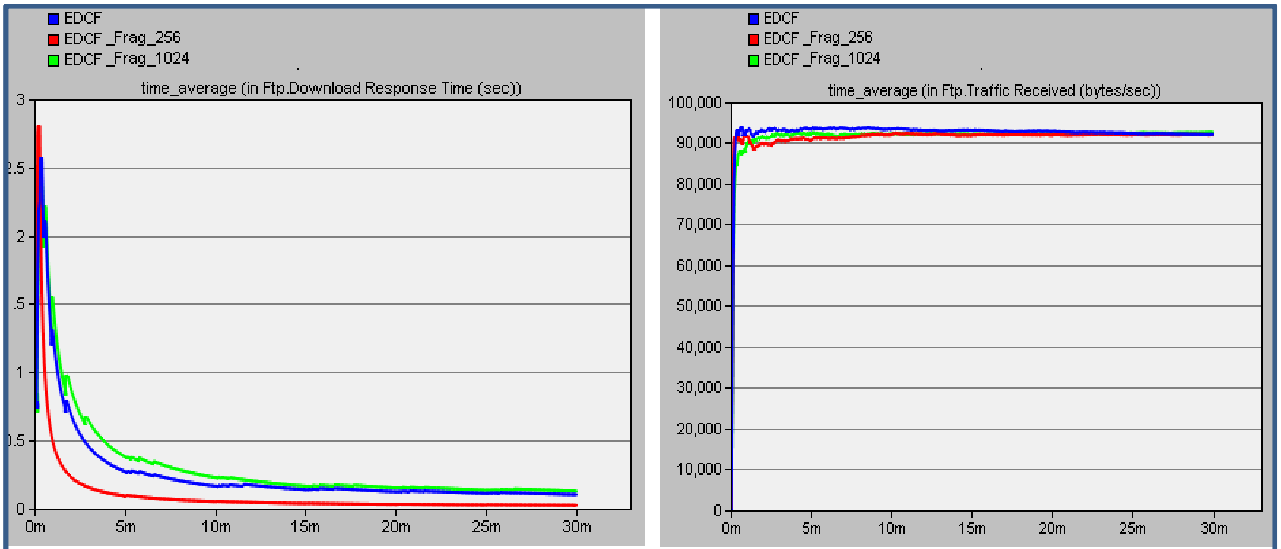


(c) MOS

Figure 67: VoIP Performance of EDCF with increased AP buffer

ii. FTP Performance

The increased buffer size did not seem to bring any additional value to EDCF ability to carry FTP when there were an equal number of VoIP and FTP stations. This section re-examines the performance of EDCF with elongated buffer length when there is a greater number of FTP stations than there is VoIP stations. It is observed in Figure 68(a) that the 256KB buffer does indeed improve the FTP download response time materially. However, no change is observed with regards to FTP traffic throughput in Figure 68(a).



(a). FTP Download Response Time (b) FTP traffic throughput

Figure 68: FTP Performance for lengthened AP buffer size

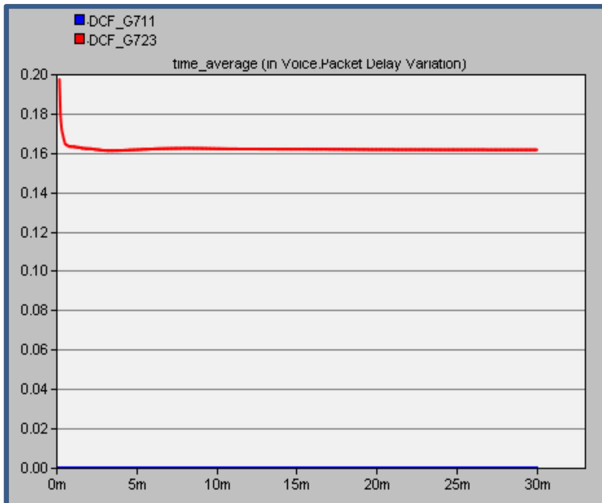
6.3. CODEC G.723

The previous sections did not evaluate the significance of codec choice. The investigations were run with the G.711 coded which presents the best MOS value but also the highest data rate Table 12. This section presents a short but different dimension to the investigation by re-evaluating VoIP performance of the three coordination schemes as per sections 6.1.1 and 6.2.1 without fragmentation, without lengthening the Access Point buffer (keeping it at 64 KB) while allowing all stations to participate in the PCF Contention Free Period. In addition this was only investigated for the case where there is an equal number of VoIP (30) and FTP (30) stations. The G.723 with 5.3 Kbps data rate was chosen because this is the lowest data rate of all codecs whist a sufficient 3.8 MOS, see Table 12. This investigation gives a snapshot of the lower data rate end extreme. The performance is compared to the G.711 as per sections 6.1.1 and 6.2.1.

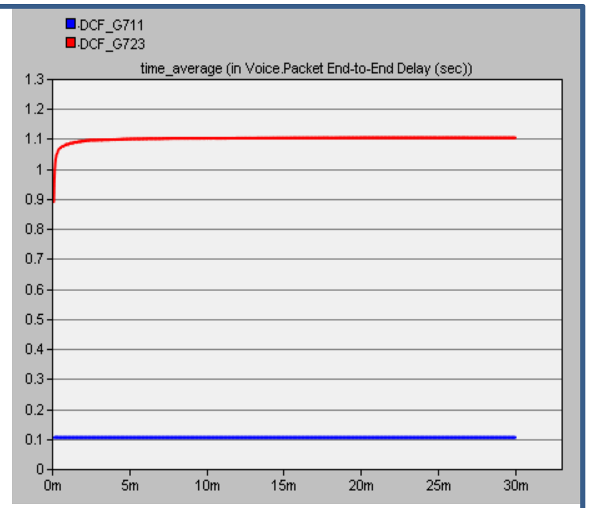
6.3.1. DCF

Figure 69(a), (b) and (c) compares the performance of DCF when it is used with G.723 and when it is used with G.711. The comparison is achieved by evaluating the packet delay variation, packet end to end delay and MOS for DCF.

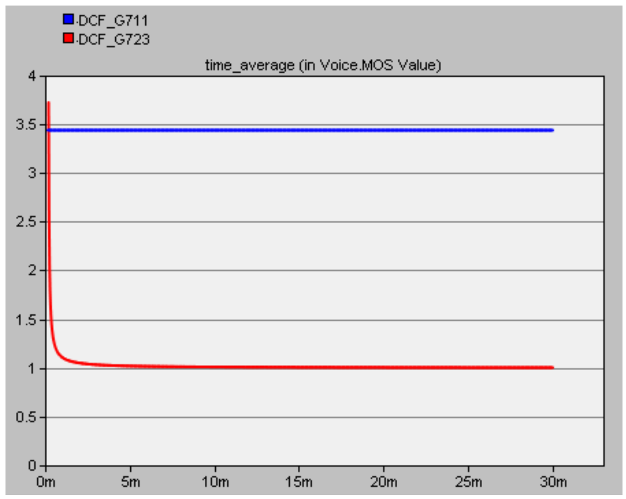
It is evident from the figure that no DCF performance improvement can be attained by using a lower rate codec to support VoIP services. This is because the packet delay variation increased Figure 69(a), packet end to end delay increased Figure 69(b) and as expected the MOS also increased Figure 69(c). Therefore, codec improvements bring no additional value to VoIP when DCF is employed.



(a). Packet delay variation



(b) Packet end to end delay



(c) MOS value

Figure 69: VoIP performance of DCF for G.711 vs G.723

6.3.2. PCF

Unlike DCF, it is observed below that the VoIP services can be improved by using a lower rate codec. The packet delay variation was degraded as a result of using a lower rate codec. However, the end to end delay improved by 43%. This is somewhat expected because the lower data rate codec results in compressed VoIP packets which in turn traverse the wireless medium faster, as a result more packets can be relayed during the Contention Free Period. The MOS was also improved from 3.49 to about 3.55, Figure 70.

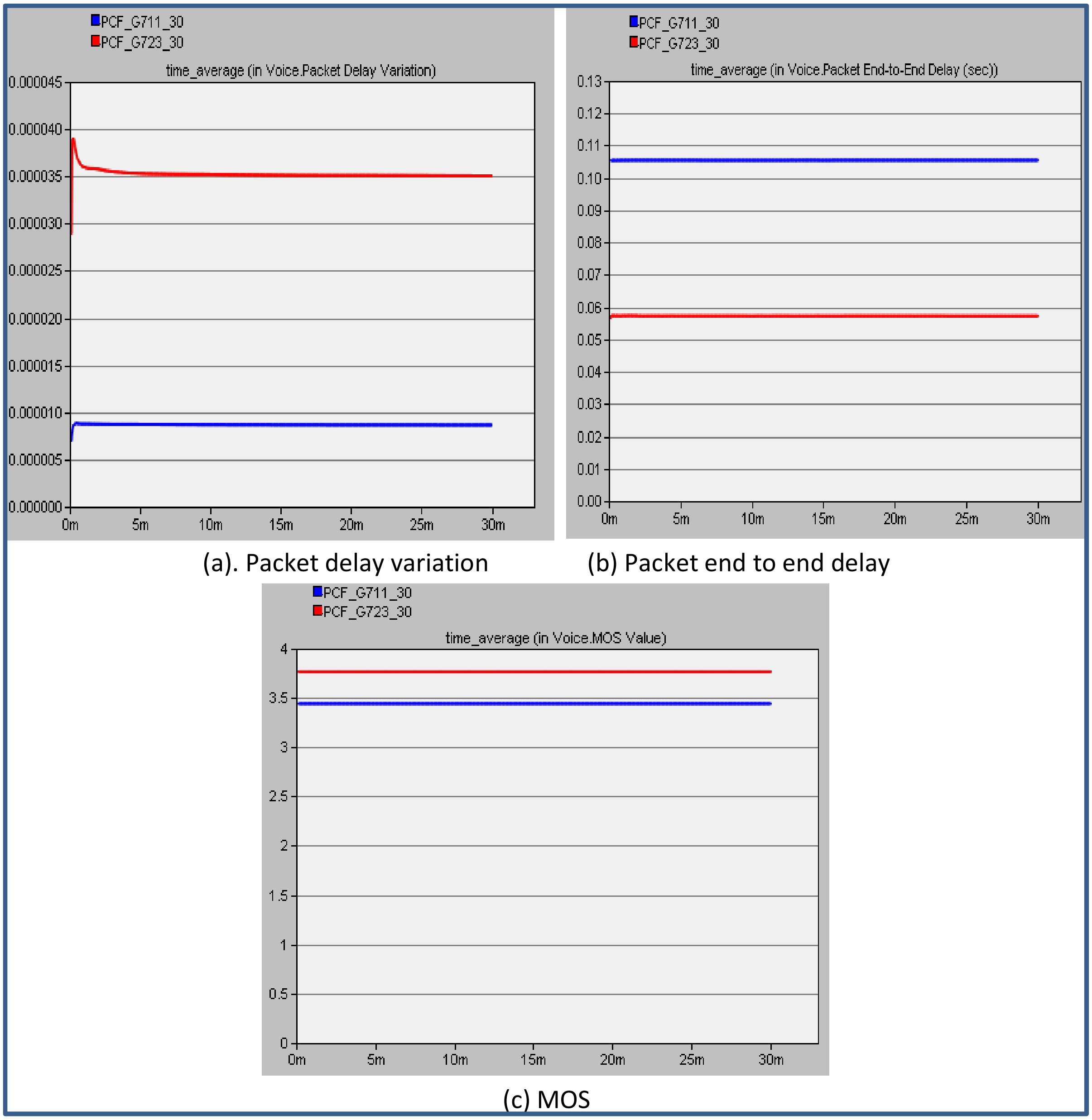
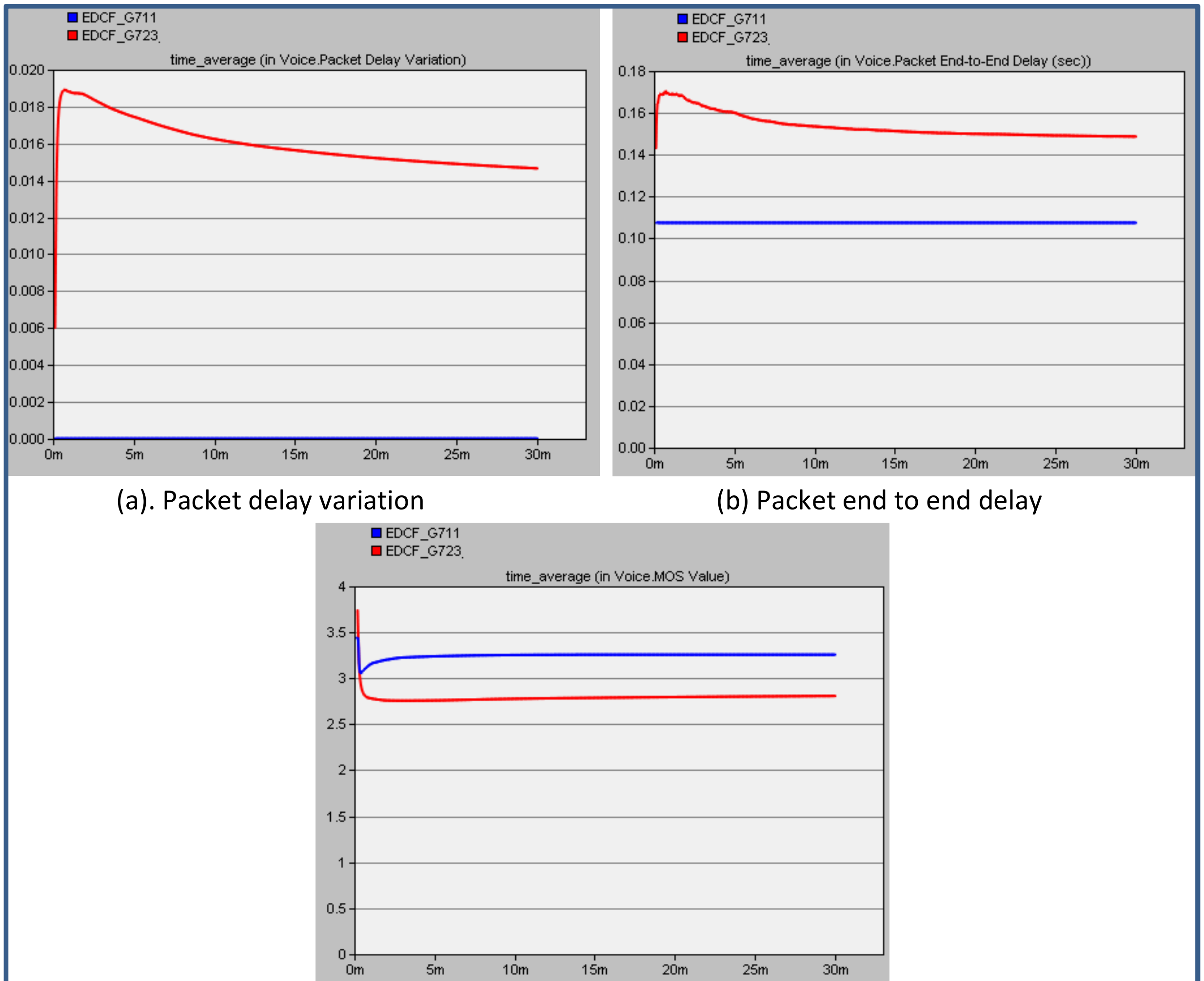


Figure 70: VoIP performance of PCF for G.711 vs G.723

6.3.3. EDCF

The Figure 71 shows that similar to DCF, the VoIP performance of EDCF is degraded by introducing a lower data rate codec. The packet end to end delay variation increased Figure 71(a), the packet end to end delay increased Figure 71(b) and consequently the MOS also increased Figure 71(a). It is also noted that the MOS was not as severely degraded as it was DC



(a). Packet delay variation

(b) Packet end to end delay

(c) MOS

Figure 71: VoIP performance of EDCF for G.711 vs G.723

6.4. Key Findings

The findings from the investigations above are summarised as follows:
When the BSS is equally loaded with FTP and VoIP stations:

- DCF gives the best MOS of 3.49 but there is no difference between the three schemes with respect to FTP experience.
- Performance of PCF with respect to VoIP or FTP cannot be improved materially by lengthening the Contention Free Period and allowing only VoIP stations to transmit during this period.
- Fragmentation does not materially improve the performance of any of PCF, DCF and EDCF with respect to VoIP or data. Although a small improvement is achieved for EDCF, this improvement is too small to justify additional processing to fragment the layer-2 frames. As was mentioned fragmentation earlier introduces additional overheads to each frame. Therefore it may potentially lead to inefficient use of the channel. It appears that the negative effects of fragmentation may be over cancelling out the positives.

The Access Point buffer was identified as a potential bottleneck. On investigating the possibility of eliminating this bottleneck by increasing the AP buffer length it was found that:

- Increasing the Access Point buffer length when PCF or DCF is in use improves the FTP download response time. However, this comes at the expense of degraded VoIP experience. VoIP users are usually more demanding that their FTP counter parts.
- Increasing the Access Point buffer length when EDCF is in service improves the MOS slightly. It also has no effect on the FTP traffic. The improved MOS is a welcome improvement. Because the cost of memory has come down over the years, an additional buffer length will not significantly increase the cost of a Wi-Fi card.

When the BSS is dominated by FTP stations the findings are summarised as:

- DCF gives the best overall performance for both and FTP, without any MAC layer enhancements
- PCF with a longer CFP did not provide any improvement in performance. However, lengthening the CFP when there is a greater number of FTP stations seem to have no effect of FTP download response time as was the case in an equally loaded network.
- The performance of EDCF with respect to VoIP worsens when there are a greater number of FTP stations.
- Employing fragmentation with PCF improved the FTP download response time and the VoIP end to end delay variation.
- Employing fragmentation with DCF degraded the FTP response time, but improved the FTP traffic throughput materially, from 30Kbps to about 93 Kbps.
- Employing fragmentation with EDCF does not improve the FTP nor the VoIP experience.
- Increasing the length of the Access Point buffer degraded the PCF VoIP and FTP data experience. The MOS and the overall FTP traffic throughput both decreased.
- Increasing the buffer length when DCF is in operation increased the VoIP delay variation but had no effect on the MOS. It also improved the FTP download response time.
- Increasing the buffer length of the Access point beyond 258 KB when EDCF is in operation did not improve performance further. Although an increase to 256 KB improved the packet delay variation and the FTP download response time.

Codec:

- Reducing the codec rate only improved the performance of PCF with respect to carrying VoIP traffic. The packet delay variation, the MOS and the packet end to end delays were improved.
- The performance of Both EDCF and DCF was severely degraded when a lower rate codec was deployed. This result is a bit surprising because the expectation is that a lower data rate codec should improve performance. This requires further investigation as it may be related to the fact that the buffer size was kept at 64 Kbps as such resulting in packet drops or packet delays and the investigation was not extended to larger buffer lengths. Unfortunately, the freely available OPNET package that was used in this study does not provide the function to measure packet-loss.

7. Chapter 7: Conclusions and Recommendations.

7.1. Conclusion

This research set-out to investigate the effectiveness of the current Wi-Fi/IEEE 802.11 MAC schemes (DCF, PCF and EDCF) with respect carrying VoIP and FTP traffic simultaneously. The investigation was carried out for a scenario where there is an equal number of both VoIP and FTP traffic as well as a scenario where there is more FTP stations competing for the same wireless resources. When the performance of PCF was not convincing, a possible enhancement, by lengthening the CFP and allowing only VoIP stations to participate was assessed. Moreover, for the three schemes the two possible Medium Access enhancements in frame fragmentation and lengthening the Access Point buffer were examined. Furthermore, the effect of codec rate was investigated by assessing a lower rate codec.

This work concludes that when there are an equal number of VoIP and FTP data stations the conventional DCF gives the best performance for VoIP. Furthermore, when the BSS is dominated by the FTP stations all schemes have nearly the same performance except that DCF still gives the best MOS. Therefore, in their current form both PCF and EDCF which were designed to provide priority to VoIP traffic do not appear to bring additional improvement to DCF when there is an equal number of VoIP and FTP traffic. Moreover, compared to a situation where there is an equal number of FTP and VoIP traffic, the performance of EDCF worsens when there are a greater number of FTP stations. Therefore, EDCF in its current form is unsuitable for carrying VoIP traffic together with FTP traffic.

This work further concludes that the length of the Contention Free Period and allowing only VoIP stations to participate during the CFP does not appear to improve PCF performance. This suggests that PCF in its current form with the possible enhancements is not sufficient for carrying VoIP and FTP simultaneously when there is an equal number of VoIP and FTP stations.

The possibility of introducing fragmentation as a MAC layer enhancement was also investigated. It was found that fragmentation can improve the FTP performance of both DCF and PCF when there are a greater number of FTP stations, but it brings no additional value to EDCF. Fragmentation introduces additional overheads because each fragment must be appended with all the headers. Therefore, larger fragments suffer from the same effects as no fragmentation in that if the fragment is corrupted it must be retransmitted. On the other hand smaller fragments suffer from additional overheads and inefficiency.

The effect of increasing the buffer size of the Access Point was investigated as a possible enhancement to the current MAC schemes. Increasing the Access Point as an enhancement to the current schemes was found to be effective with EDCF as it improved the MOS. It was also found to improve the FTP response time of both PCF and DCF when there is an equal number of FTP and VoIP stations as well as in an FTP dominated network. Therefore, varying the buffer length of the Access Point can potentially improve the current MAC schemes.

Lastly the effect of a lower rate codec was investigated as a supplement to the current MAC schemes. The lower codec rate was found to improve performance of PCF, but severely degraded the performance of DCF and EDCF. It is concluded that the VoIP codec choice does affect the performance but it cannot be viewed in isolation.

In conclusion the research question as to whether the current MAC schemes are sufficient for carrying VoIP and FTP, all the three schemes achieved a MOS between 3 and 3.5 with the G.711 and the FTP throughput of between 60Kbps and 90Kbps. The VoIP performance is somewhat dissatisfactory for most users. However, a brief investigation showed that the MOS can be improved to well above 3.5 but still below 4 with a lower rate codec for PCF but not for any of EDCF or DCF. Therefore, a further study of the various codecs needs to be carried out to assess the effect of codec choice. Moreover, the available MAC enhancements, fragmentation and Access Point buffer increment do not appear to bring any significant improvements to the performance of any of the three schemes.

7.2. Recommendations

Firstly, it was found that when the codec scheme with a lower data rate was used, the performance of DCF and EDCF was degraded. Further work should investigate how the lower codec rate responds to an increased buffer length. A further investigation into the different codec schemes needs to be carried out in order to rank the codecs when voice is carried over Wi-Fi in the presence of FTP data. This will give some insight into a framework for choosing codec schemes along with the MAC schemes in order to support voice services on the same BSS with FTP services.

Secondly, in this investigation, it was found that lengthening the CFP and allowing only VoIP station to participate does not materially improve the voice or FTP experience. Therefore is may be worthwhile to investigate a scenario wherein the DTIM periodicity is reduced. In other words, setting the DTIM to a value that is lower than 0.02s in order to allow more frequent Contention Free Periods for servicing voice stations.

Furthermore, it was assumed that there are no silent Voice intervals in order to strain the MAC schemes. This investigation may be repeated by considering the behaviour when silent intervals are introduced. Although the objective was to test a worst case scenario, in reality there are silence intervals.

The Access Point was identified as a potential bottleneck, but increasing the buffer size to relieve the bottleneck did not appear to improve performance materially. In addition the performance of both EDCF and DCF may be re-evaluated by increasing the Transmission Opportunity (TXOP) of the Access Point relative to other stations in the network.

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