Final Honours Project Report

"Performance analysis of VoIP in an IEEE 802.11g WLAN deployment"

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Abstract

There has been a gradual increase in the implementation of 802.11g Wireless Local Area Networks (WLANs) over the past few years due to increased bandwidth availability over 802.11b networks, flexibility and decreasing equipment costs. Wired LANs were the original source for Voice over IP (VoIP) however the implementations of VoIP on WLANs have rapidly increased. 802.11g WLANS can offer actual throughput of up to 22mbps which is approximately a quarter of what a wired 100mbps can offer, therefore providing much less bandwidth than a traditional wired LAN. With many applications competing for bandwidth in an office environment such as Internet access, FTP, Email, VoIP performance can be seriously degraded due to the lack of bandwidth available.

There are many VoIP codecs available which can be implemented, with each codec utilising a specific amount of network bandwidth. The choice of codec to be implemented can have a major effect on the speech quality which is received and also on the maximum number of simultaneous 2-way calls possible.

This study analysed the performance of common VoIP codecs in an IEEE 802.11g WLAN deployment to identify which codec would allow the highest number of simultaneous 2-way voice calls whilst retaining acceptable speech quality of a Mean Opinion Score (MOS) equal to or greater than 3.5. Audio payload was also increased from 20ms to 30ms to analyse whether this had an effect on codec performance.

From the results gathered, it was found that the G.729 codec allowed the maximum number of simultaneous 2-way voice calls possible in an IEEE 802.11g WLAN deployment for both 20ms and 30ms audio payloads. The increase in number of calls by increasing the audio payload was a small increase.

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1.0 Introduction

1.1 Background

Voice over Internet Protocol (VoIP) is the technology used which transmits voice conversations over IP packet switched networks. VoIP was initially deployed on Wired LAN's however the recent growth of Wireless Local Area Network (WLAN) technology has seen VoIP solutions being implemented on these as they allow users greater flexibility and mobility (Dini, Font-Bach & Mangues-Bafalluy 2008). There are many factors to consider when implementing a VoIP solution, as each will have an effect on performance (Haugdahl 2005).

WLAN's benefit over wired networks for many reasons (Xu 2003); they offer scalability and mobility, along with long and short term cost savings, whilst also having the ability to provide network coverage in areas where it is not possible for wired networks to reach. However the data rate provided by an 802.11g network is 54 Mbps whilst actual throughput is up to 22 Mbps (Hucaby 2006). This throughput is considerably less than the 1000Mbps which can be achieved by a modern wired network thus questioning the efficiency of WLANs for carrying voice data (An Chan, Soung Chang Liew 2007). Background traffic such as web surfing, email and other data applications can also deteriorate VoIP (Wei Wang, Soung Chang Liew & Li 2005).

In order for a VoIP conversation to take place, an algorithm is used to encode and then decode the voice which is heard. A three step operation is performed for analog to digital voice conversion. The first step is to sample the analog signal, which is then quantized into a binary expression (step two). In the final step, the samples are compressed to reduce bandwidth. The algorithm which is used to perform this operation is called a codec (CiscoWiki). Some codec's compress less than others therefore offering higher call quality, but by doing so, consume more bandwidth, whilst other codec's reduce bandwidth by using higher compression which can result in packet loss, delay and jitter (Haugdahl 2005, Haugdahl 2005), all of which contribute to the degradation of speech quality(Narbutt, Davis 2005). The codec's which will be tested are of ITU-T standard(Sinnreich, Johnston 2006).

The amount of bandwidth consumed can also be affected by the audio payload size (Hole, Tobagi 2004). The audio payload is the amount of voice which is inserted into a packet and can be represented using bytes or milliseconds (ms) (OzVoIp). It is possible to reduce the number of packets required to transmit a call by increasing the audio payload whereas decreasing the audio payload will require more packets to transmit the same call (Garg, Kappes 2003). Decreasing the number of packets can reduce the amount of headers on the network which can improve the availability of bandwidth (Melvin, Murphy 2002) however using a larger audio payload can also result in delay as it actually increases the length of time for the construction of a single packet(Ahmed, Jiang & Horiguchi 2008).

Furthermore, wireless data is vulnerable to interception by anyone who is in close proximity to the network range. A number of steps have to be taken to ensure that wireless data which is being transmitted is encrypted. Introducing wireless security adds overhead to voice packets which can increase bandwidth utilisation therefore having a degrading affect on call quality(Wireless Attacks Primer).

A method of assessing communication quality is by using the Mean Opinion Score (MOS) (Itoh, Tajima & Kuwabara 2000). The MOS is a numerical indicator (Systems,Inc Sports Team Analysis and Tracking, Systems 2003) of the quality of received speech after it has undergone transmission and compression. It is indicated using a single number with 1 being the poorest quality, and 5 being the best (Lingfen Sun, Ifeachor 2002). A Mean Opinion Score of 3.5 is deemed as acceptable (Raake 2006). It is possible to obtain the Mean Opinion Score of a phone call using the the codec, jitter, delay and packet loss as variables for the calculation (Meddahi, Afifl & Zeghlache 2003) or by using the E-Model tool(Passito et al. 2005).

Using the aforementioned issues, the purpose of this project will be to determine which codec can allow the highest number of simultaneous 2-way VoIP calls in an 802.11g WLAN deployment. By using different codec's; delay, packet loss and jitter and Mean Opinion Score will be recorded which will give a numerical indication of whether the speech quality is acceptable when multiple calls are in progress. The effect on call quality will also be recorded after increasing audio payload sizes on the codecs used.

1.2 Project Outline & Research Question

1.2.1 Project Method

An experiment will be conducted to measure the variable outcomes of the voice calls taking place. These are delay, jitter and packet loss. A value for the Mean Opinion Score can also be obtained from the network simulation tool.

Research Question

In order to accomplish the aim of the project, and to focus the experiment, the following research question will be answered:

"Which common Voice over IP codec will allow highest call capacity of simultaneous 2-way phone calls to be made in an IEEE 802.11g WLAN deployment whilst retaining a Mean Opinion Score of greater than or equal to 3.5?"

Call capacity will also be measured to identify the affects of increasing audio payload size.

The purpose of the following section identifies the objectives which will help to give direction to the project and will help investigate the areas that will be involved during the experiment.

Aims & Objectives

The aim of this project is to determine which VoIP codec allows the highest number of simultaneous 2-way phone calls in an 802.11g WLAN deployment, whilst retaining acceptable speech quality. The findings from the experiment will provide the reader of this study a clear understanding on which VoIP codec is most suitable.

Through a literature review, the following objectives will be met:

1. Examine the various common VoIP codec's available and how their attributes differ from one another.

This will identify the various VoIP codec's available. The information found will be useful as codec's key characteristics will play a vital part when tested in the experiment.

2. Investigate common wireless networks available.

The findings from this objective will provide an understanding of the various types of wireless networks available. The information from this research will be useful as it will identify which setup will be most suitable for conducting the experiment.

3. Identify the variables which affect speech quality, and what are deemed to be acceptable values.

This will reveal details on the factors which affect quality of service in VoIP deployments on WLANs. The findings from this objective will identify the variables which are used in order to determine speech quality.

Hypotheses

Hypotheses have been constructed of what may be the possible outcomes of this particular project. Each hypothesis has been concluded from thorough literature research and has a justification which highlights the fact that investigation into the particular area is needed.

- The G.729 codec will allow higher call capacity whilst retaining acceptable speech quality.

The G.729 codec utilises much less bandwidth than the G.711 codec at the expense of speech quality however it can still retain a Mean Opinion Score of 3.5 whilst allowing a greater number of simultaneous 2-way voice calls to take place.

- An increase in audio payload size will increase the maximum number of simultaneous 2-way voice calls possible.

Increasing audio payload increases the size of the voice packets which are transmitted therefore less packets are sent. This reduces the amount of bandwidth being utilised however any packet loss will severely degrade speech quality as more voice will be lost.

2.0 Literature Review

The aim of the literature review is to complete the objectives which are set out in section 1.2.1. By completing the objectives in the previous section, a high level of understanding on VoIP technology will be revealed by the literature review in the following areas:

- Fundamentals of VoIP technology
- Common VoIP codec's
- Typical WLAN solutions
- VoIP Quality of Service

2.1 VoIP – The Fundamentals

Voice over Internet Protocol (VoIP) is the technology used which transmits voice conversations over packet switched networks. Over recent years many organisations have started to implement VoIP solutions to replace their existing traditional telephone networks. The two main reasons why this change takes place are due to cost reduction and flexibility. A substantial amount of equipment is needed to host a separate voice network thus deploying a VoIP solution which can take advantage of the data network already in place can greatly reduce equipment and operations costs (Wan, Hui 2005).

IP networks were not designed with voice in mind, thus problems may occur when implementing Voice into an IP based network (Raake 2006), especially a WLAN deployment which has considerably less bandwidth to offer compared to a common 100Mbps wired network, or even modern gigabit wired networks which offer up to 1000Mbps bandwidth. An 802.11g WLAN will offer 54Mbps whilst actual throughput is up to 22Mbps(Davidson, Peters & Gracely 2000).

To function efficiently, voice traffic requires low delays. It is recommended that no more than 150ms of one way, end-to-end delay should occur to retain good voice quality(ITU-T Recommendation G.107 2003).

2.1.1 Protocols

Voice conversations travel across medium through the use of various protocols. They can be classified into Transport Protocols and Call Signalling Protocols.

2.1.1.1 Transport Protocols

Voice data is transmitted across the media by using the Real-Time Transport Protocol (RTP), and the RTP Control Protocol (RTCP)(Sfairopoulou, Macian & Bellalta 2006). RTP provides a standardised packet format for delivering video and audio over the Internet, however it does not guarantee the delivery of data. RTCP is a sister protocol of RTP as it provides the control of packet flow for individual RTP streams. The main function of RTCP is to obtain statistics on quality aspects of the media distribution during a particular session. The data is then sent to the session media source and to any other participant. (Ranjbar 2007).

There are other transport control protocols available such as the Transport Control Protocol (TCP) and the User Datagram Protocol (UDP). These are the two main transport layer protocols which are used on modern IP networks however they are not suitable for carrying voice data. A technique called positive acknowledgement with retransmission is used by TCP to ensure the delivery of data packets. If a packet is lost during transmission, it is retransmitted, which seems fine for data but not suitable for carrying conversations.

UDP uses a simple transmission procedure, and does not include any sort of handshaking or retransmission procedure for guaranteeing the delivery of packets. This means it’s more suited to real time transmissions due to the fact that it includes less overheads however the main drawback of UDP is that it does not provide flow control or synchronisation between peer devices.

2.1.1.2 Call Signalling Protocols

Call signalling protocols are the most frequently referenced VoIP protocols. The purpose of the call signalling protocols is to establish a connection between the endpoints.

The two most popular protocols available are H.323 which is of ITU standards and Session Initiation Protocol (SIP) which is and Internet Engineering Task Force (IETF) standard (IETF).

H.323 was approved by the ITU in 1996. It provides a base for video, audio and data communications across IP based networks. Messages appear in binary format as it was initially based on traditional signalling protocols from the PSTN thus making it very complex(Goode 2002).

SIP was developed and approved by the IETF in 1999(Sinnreich, Johnston 2006). The advantages of SIP over H.323 are that SIP is transport independent and can be used with UDP and TCP, thus making it usable in various other networks. SIP messages also appear in clear text making it relatively easier to understand (Geier 2007).

For the purpose of the simulation, the SIP protocol will be implemented as it’s less complex than the H.323 protocol, however research shows that performance difference is relatively minimal(Sfairopoulou, Macian & Bellalta 2006).

2.1.2 VoIP Codecs

Various codecs are available to use in VoIP solutions. A codec is an algorithm which is used to encode and decode the voice which is heard. Voice as we hear it is analogue, therefore conversion to a digital format is required so that it is prepared for transfer over the IP network. Once it reaches the other end, it then needs to be decoded so the person can hear what is being said. There are various methods for encoding and decoding, many of which require some sort of compression to reduce the bandwidth of the conversation. Most VoIP devices utilise codecs which are standardised by the International Telecommunication Union – Telecommunication Standardisation Sector (ITU-T) as they are interoperable between various vendors. Two of the most common VoIP codecs in use today are the G.711 and the G.729 codecs(Garg, Kappes 2003).

2.1.2.1 G.711 Codec

The G.711 codec provides precise speech transmission and uses the µ-Law and A-Law algorithms. The µ-Law algorithm is used in Japan and North America while the A-Law algorithm is used in Europe and the rest of the world. Both algorithms are logarithmic however the A-Law uses less processing power(Haugdahl 2005).

The G.711 codec does not perform any compression and requires a fairly low usage of computational power thus it is able to deliver a very high quality of speech. It also produces small sized packets which in turn reduce the overall delay however one drawback of this codec is that it requires a fairly high bandwidth of 64kbps (8000 samples per second at 8 bits each) which when compared to other common codecs, is high(Dini, Font-Bach & Mangues-Bafalluy 2008). Previous research carried out by Narbutt (2005) showed that the G.711 codec gave higher user satisfaction with regards to speech quality over a WLAN deployment. The study analysed the performance of the G.711 and G.729A codecs however it did not take into account the maximum call capacity achievable by the two codecs.

2.1.2.2 G.729 Codec

The G.729 codec involves compressing audio, which means that it consumes considerably less bandwidth than the G.711 codec. Compared to the 64 kbps used by G.711, G.729 uses only 8 kbps. Voice samples are collected for eight times longer than the G.711 codec which may lead to considerable delays. A study by Hole (2004) shows that the G.711 codec effectively optimises the bandwidth used per connection however it requires high processing power thus allowing greater call capacity than the G.711 codec at the expense of voice quality.

There are other VoIP codecs available such as G.723.1, G.726 and G.728 however a number of previous studies have compared the difference between G.711 and G.729 as they are the most common available due to one providing excellent speech quality and the other consuming less bandwidth (Ding et al. 2007). The performance of the G.711 and G.729 codecs will be analysed during the simulation for this study.

2.1.3 Bandwidth Requirements

There are many factors which affect how much bandwidth will be consumed by a single VoIP call. Although the G.711 and G.729 require 64kbps and 8kbps of bandwidth respectively, this is not an indication of the actual bandwidth consumed as there are overhead costs involved. A voice packet consists of IP, UDP, RTP headers along with the audio payload(Wallace 2008).

The audio payload is the amount of voice which is inserted into a single voice packet. By default, the G.711 and G.729 codecs insert 20ms of voice however for the G.711 codec, 20ms of voice consumes 160 bytes of the packet whereas 20ms of voice in the G.729 codec uses just 20 bytes(Hole, Tobagi 2004).

Increasing the audio payload can have both positive and negative effects on call quality. The ratio between header and payload is increased when increasing the size of the audio payload, therefore reducing the number of packets that need to be transmitted which in turn consumes less bandwidth meaning higher call capacity. However, more audio is lost if packet loss occurs which reduces speech quality at the receiving end(Hole, Tobagi 2004). Garg & Kappes (2003) analysed the performance of G.711 and G.729 and results showed that by increasing audio payload from 20ms to 30ms can increase call capacity however as mentioned earlier, it can have a negative effect on delay, jitter and packet loss.

The size of the IP header in a voice packet is 20 bytes, whilst UDP is 8 bytes and RTP is 12 bytes, giving a total of 40 bytes.

The diagram below gives a visual indication of a voice packet using the G.729 codec.

20	8	12	20
IP	UDP	RTP	G.729 Audio Payload

Figure 1 - G.729 Voice Packet

Using the following formula, the total bandwidth required per call can be calculated:

Bandwidth = Total Packet Size x PPS

PPS is the number of packets that need to be sent every second to deliver the specified codec bit rate:

PPS = Codec Bit Rate / Voice Payload Size

Using the IP, UDP, RTP and Audio Payload values mentioned above, the bandwidth required for one call can be calculated:

PPS	=	8 kbps / 160 bits
	=	50 pps
Total Packet Size	=	IP + UDP + RTP + Audio Payload
	=	20 + 8 + 12 + 20
	=	40 bytes
	=	320 bits
Bandwidth:	=	Total Packet Size x PPS

= 320 x 50
= 16 kbps

It can be seen above that by adding various overheads, the bandwidth requirements has doubled the 8kbps which the G.729 codec consumes. The bandwidth per call increases further by adding Layer 2 Data-Link overheads which were not taken into account when calculating total packet size, along with basic wireless security which again, was not taken into account(Cisco).

The size of a voice packet can be reduced greatly by implementing a Compressed RTP (cRTP) header. This replaces the IP, UDP and RTP headers and replacing them with a 2 byte header if a checksum is not used, or a 4 byte header if checksum is used. An experiment carried out by Xiao et al (2004) showed that a 25% bandwidth saving was achieved on an 802.11 WLAN using cRTP however this will not be implemented during the experiment as it is only recommended on links which have less than 2 Mbps of bandwidth available(Cisco).

2.1.4 Quality of Service (QoS)

Networks generally tend to operate on a best-effort basis where all traffic has the same priority and the chances of being delivered in a timely manner are equal. This can also be interpreted as that the chances of packets being dropped when network congestion occurs are also equal(Li, Cui 2006). In networking alone, quality can mean a number of things however when it concerns speech, it means being able to listen to a clear and uninterrupted voice. The three factors which affect quality of received speech are delay, packet loss and jitter(Geier 2007). Quality of Service can be implemented using various congestion management tools such as priority queuing (PQ), custom queuing (CQ), Flow-based weighted fair queuing (WFQ) and also queue management tools such as Random early detection (RED) and WRED however as there will be no background traffic competing for bandwidth space during the experiment, QoS will not be implemented.

2.1.4.1 Delay

Knowledge of delay components in networks when designing networks which carry voice packets is vital. If all potential delays are accounted for, it helps maintain acceptable network performance. In terms of delay, anything below 150 ms is deemed acceptable. When delay extends beyond 150 ms then holding a conversation becomes increasingly difficult. There are two variations of delays; fixed delays, and variable delays(Narvaez Diaz, Perez Diaz 2006).

2.1.4.1.1 Fixed Delays

- Coder Delay: The time taken for voice signals to be converted from analog to digital signals. This is also known as processing delay. The delay tends to vary depending on which codec is used and also depends on processor speed.

- Packetisation Delay: The time taken to fill IP packets with digital voice. This can also be known as accumulation delay as the samples of digital voice accumulate in a buffer before being released.

- Serialisation Delay: The amount of time required to clock a voice frame onto the network interface. It directly relates to the clock rate which is set on the trunk.

- De-Jitter Delay: This operation transforms variable jitter delay into a fixed delay as it holds the first sample received for a length of time before being played out. The holding operation of the de-jitter buffer is known as the initial play out delay.

2.1.4.1.2 Variable Delays

- Queuing/Buffering Delay: Queuing delay is the length of time a packet is queued for transmission across the network. Voice must have priority in the router therefore the queued frame should only wait for a data frame that has already been transmitted, or for other similar packets which are ahead of it in the queue.

- Network Path Delay: This represents the amount of time which it takes for packets to travel across the network connection(Cisco).

2.1.4.2 Packet Loss

The percentage of packets lost during transmission whether it's data or voice. Packet loss tends to occur when a high amount of data is travelling across the network, causing congestion and thus packet loss. Packet loss should ideally be under 1% (when using the G.711 codec) or speech quality will be degraded, and in some cases, voice calls may be dropped, however when using the G.729 codec, packet loss should be kept below 1%(Geier 2007). There are various methods used to conceal the affects of packet loss such as Packet Loss Concealment (PLC). This method depends on the type of codec used however it works by replaying the previously received packet at a higher attenuation. Research has shown that packet loss becomes increasingly difficult to conceal when it occurs in bursts(Wallace 2008)(Hneiti, Ajlouni 2006).

2.1.4.3 Jitter

When packets are transmitted from the source, they are sent out at a steady rate however when they reach the destination, they tend to arrive with varying time differences between consecutive packets. The cause of this is due to queuing and high network traffic. The average time between two consecutive packets is called Jitter. To combat this, a de-jitter buffer is used as explained in section 2.1.4.1.1. The de-jitter buffer holds packets for a length of time so they can be played at the same rate in which they are transmitted. When a high level of jitter occurs, the packets are held in the de-jitter buffer for a greater period of time, thus adding to the accumulating delay which should be kept below 150ms(Dao, Malaney & Exposito 2005). The delay for the jitter itself should be no more than 30ms(Lingfen Sun, Ifeachor 2002).

2.1.5 Assessing Speech Quality

Voice quality is directly affected by the three factors mentioned above; delay, packet loss and jitter. A method of assessing quality of service for VoIP is by measuring speech intelligibility at the end point of a voice call. One way of doing this is by using The Mean Opinion Score (MOS). The MOS is a subjective method of testing speech quality. There are a number of other tests available to measure speech quality. Perceptual Speech Quality Measurement (PSQM) and Perceptual Evaluation of Speech Quality (PESQ) are two other methods of assessing speech quality. Both are objective methods of assessing speech. Both methods provide an automated test operation thus negating the need for human interpretation for calculations; however PSQM does not take into account the effects of packet loss and jitter as it was originally designed to test circuit-switched networks. Objective methods generally tend to be expensive and complex to conduct(Haugdahl 2005) therefore the quality of speech which will be analysed for this study will be determined by the MOS.

2.1.5.1 Mean Opinion Score

The quality of a voice call can be measured using the Mean Opinion Score (MOS). The MOS test is specified by ITU-T recommendation P800.1(ITU-T Recommendation P.800 1996). The test provides a numerical indication of the perceived quality of received media after transmission and compression, and is expressed as a single number in the range of 1 to 5 where 1 is lowest quality and 5 is the highest. The disadvantages of using this method is that it requires a large number of listeners and can be rather time consuming however to combat this, a number of software applications are now available to obtain MOS readings. Although they do lack the human input, they do take into account all factors which can influence the MOS. A MOS value of 3.5 is the minimum value of acceptable speech quality, therefore 3.5 is used as the reference value for this study(Raake 2006).

2.1.5.2 The E-Model

The E-Model is an objective method of evaluating perceived speech quality in VoIP systems(http://www.ixiacom.com/library/white_papers). The score obtained from the E-Model calculation is the transmission rating R factor which ranges from 0 to 100 with 100 being excellent. An R factor value of below 60 is not recommended (Passito et al. 2005). The value for the R factor can be obtained by using the following equation:

$$= 93.4 - \quad (\quad) - \quad (\quad , \quad)$$

The signal to noise ratio is represented by the 93.4 value. is a representation of the one way delay and is a function of the codec used and the packet loss rate.

Once the R value is obtained, an MOS value can then be calculated using the following equation(Carvalho et al. 2005):

For $R < 6.5$: MOS = 1
For $6.5 \leq R \leq 100$:

$$MOS = 1 + 0.035\,R + 7.10^{-6}\,R(R-60)\,(100-R)$$

For $R > 100$: MOS = 4.5

The above equations have been used in previous studies including one carried out by Passito (2005). An E-Model implementation was used to evaluate speech quality in VoIP networks with results supporting the fact that the E-Model tool is robust and trustworthy however previous studies have also been carried out using the OPNET Modeler simulation tool which have also produced reliable results (Haq et al. 2007) with MOS values without the need for any objective method of evaluation such as the E-Model therefore the E-Model tool will not be used in this study to calculate MOS.

2.2 WLANs

Over the past few years, the popularity of Wireless Local Area Networks (WLANs) has risen dramatically. The main reasons for the increase in demand for WLANs are flexibility, cost reduction and high data rates. WLANs are able to provide network connectivity within enterprise networks in places where it was not possible to provide any sort of network connection using a wired Ethernet connection. Also, providing network connectivity in a building is as simple as adding a new wireless access point to which clients can authenticate to rather than having to pay an engineer to install new Ethernet cabling thus dramatically reducing implementation costs. Combining this with the fact that 802.11g wireless network can provide a 54mbps connection, it can be seen why many enterprise networks are making the move from wired to wireless(Medepalli et al. 2004). Actual throughput of an 802.11g network is approximately 22mbps.

The 802.11g standard has been chosen for this simulation due to the growing popularity of it. It also makes a considerable improvement on available bandwidth offered by the 802.11b standard therefore making it a perfect choice for the simulation.

The 802.11n standard which can provide actual throughput of around 150mbps has not yet been ratified (Cisco.com - 802.11n Wireless Technology Overview) therefore this will not be used in the simulation, however it can be used for further study.

2.2.1 WLAN Security

Since information which is transmitted inside a WLAN is not travelling through physical cabling, then it is vulnerable to interception by anyone who has the necessary tools and is within the network range. Wireless attacks can be split into four categories; passive, active, man-in-the-middle and jammer attacks(Wireless Attacks Primer). Wireless security must be implemented to protect against these potential threats.

2.2.2 Wi-Fi Protected Access (WPA and WPA2)

Wi-Fi Protected Access (WPA and WPA2) is a security protocol created by the Wi-Fi Alliance to secure wireless networks. WPA and WPA2 were developed in response to various vulnerabilities in the predecessor of WPA, Wired Equivalent Privacy (WEP) (Park, Dicoi 2003).

WPA2-PSK can severely hamper the chances of any hacker attempting any of the attacks mentioned above whilst at the same time not adding a great amount of overhead to the voice packets compared to other methods of security(Xiao, Zarrella 2004). Pre-Shared Key (PSK) is designed for personal or small enterprise networks which don't require the added complexity of an 802.1x authentication/radius server.

There are other security methods available such as IPSec however the main drawback is the large overhead which IPSec adds to the voice packets. IPSec comes in two modes; transport mode and tunnel mode. In transport mode, encryption is applied onto the payload of the IP packet whereas in tunnel mode, the whole packet itself is encrypted. Previous research has shown that IPSec can have a crippling affect on bandwidth utilisation as anywhere between 30 – 70 bytes can be added onto the overhead depending on which configuration of IPSec is utilised thus severely degrading speech quality(Xiao, Randhawa & Hardy 2004).

The OPNET Modeler tool unfortunately does not allow wireless traffic to be encrypted using WEP, WPA or WPA2 therefore it cannot be implemented for the experiment. OPNET does allow IPSec functionality however the purpose of the experiment is to determine speech quality related to call quantity without having to add any major overheads which can have a detrimental effect on both speech quality and call quality therefore IPSec will not be implemented during the simulation.

2.2.3 WLAN Topologies

There are three types of wireless networks available; Independent Basic Service Set (IBSS), Basic Service Set (BSS) and Extended Service Set (ESS).

2.2.3.1 Basic Service Set

A Basic Service Set topology centralises control by placing a wireless access point in the topology. Wireless clients must authenticate with the access point in order to gain network access and must satisfy the following criteria; matching Service Set Identifier (SSID), matching frequency channel and data rate, and security credentials. Below is an example of a Basic Service Set network. This is also known as an Infrastructure network.

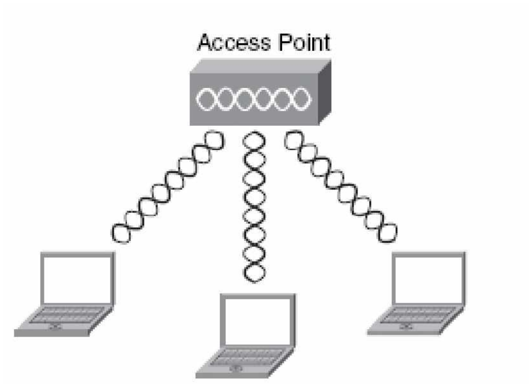


Figure 2 - Infrastructure Network

The above topology usually operates within a single building allowing wireless connectivity to clients. A similar topology will be implemented during the experiment where wireless clients will authenticate themselves with a wireless Cisco access point, similar to how the clients in the above diagram authenticate themselves with the access point.

Previous research has shown that that the Infrastructure topology is the most suitable one to simulate as it does not include the complexity of having multiple access points such as the Extended Service Set topology(Ma 2003).

2.2.3.2 Independent Basic Service Set

An IBSS network allows wireless clients connect to each other and communicate, without associating with a wireless access point. This type of topology is also known as an ad-hoc wireless network. No access point is present and the wireless clients are directly communicating with each other. This type of topology is useful if clients who are in close proximity with each other want to share files therefore since the purpose of the simulation is to evaluate call quality and call quantity, then this type of topology is not suitable for the simulation.

2.2.3.3 Extended Service Set

The third type of wireless network is an Extender Service Set (ESS). This type of network is similar to the BSS however they tend to be used when access points are based at different geographic locations. Below is an example of an ESS network.

The diagram above shows a multi layered switch placed at the centre of the network which connects together the two networks either side of it. These are basically two infrastructure networks joined together by a multi layered switch. The two access points in the topology above represent networks located at two separate geographic locations.

This topology will not be utilised during the experiment as the project is focusing on a single site topology however this can be used for further study.

2.3 Voice over WLAN – VoWLAN

Delay, jitter and packet loss must be kept to an absolute minimum if real time communications such as VoIP is to be implemented and run efficiently over a WLAN deployment. Wireless networks have less bandwidth available in comparison to wired networks thus all possible bottlenecks must be accounted for (Geier, 2007).

2.3.1 Call Capacity in WLANs

The number of simultaneous calls possible significantly decreases when VoIP is deployed on a WLAN rather than a Wired LAN due to the reduction in available bandwidth.

Previous studies have shown that the G.729 codec permits higher call capacity than the G.711 codec at the expense of call quality. A study by Hole & Tobagi (2004) shows that the G.729 codec's maximum achievable MOS is 3.65. It also states that the G.711 codec has higher speech quality due to reduced coder process delays however this is at the expense of call capacity. The study by Hole & Tobagi (2004) is conducted on an 802.11b network which only has 11mbps of bandwidth available. A 54mbps 802.11g can offer approximately up to four times the call capacity of an 802.11b network(Geier 2007).

2.4 Conclusion

The literature review has satisfied the aims and objectives set in section 1.2.1 of this report.

This literature review revealed the various elements which can combine to form an efficient VoIP deployment over an 802.11g WLAN. These elements were identified through various journal articles, previous researches, conference proceedings, white papers and manuals and therefore identified which elements were necessary for the experiment.

The literature review identified that a simulation experiment would be most suited to measure speech quality in relation to call quality. Previous research identifies that the OPNET Modeler tool can provide accurate MOS, delay, jitter and packet loss readings allowing them to be compared alongside the recommended values which were also identified during the literature research.

Also identified was the most common voice codecs in use today; G.711 and G.729. These codecs are available to use within OPNET Modeler.

Along with the above findings, a suitable WLAN topology to be constructed within OPNET Modeler was also identified thus providing the building blocks for the simulation to measure speech quality.

Below is a table which identifies the various elements of a VoIP deployment which were obtained from the literature review. The table also shows which elements were used in the experiment and also a brief justification as to why elements were not included.

Table 1 - Experiment Factors

Call Signalling Protocol:	Used In Experiment	Reason Not Used
SIP	ü	
H.323	x	More complex than SIP for this experiment
VoIP Codecs:		
G.711	ü	
G.729	ü	
G.723.1, G.726 and G.728	x	Not as commonly used as G.711 and G.729
Wireless Topology:		
Basic Service Set	ü	
Extended Service Set	x	Doesn’t represent a wireless topology in the same georgraphic location
Ad-Hoc	x	No access point is used, not suitable for VoIP calls.
Security:		
WPA2-PSK	x	Not available in OPNET Modeler
WEP	x	Very basic encryption and easily cracked. Not available in OPNET Modeler.
IPSec	x	Can have devastating effects on speech quality due to large overheads.

From the literature review, it can be concluded that the G.729 codec will provide a higher call capacity than the G.711 codec at the expense of speech quality. Using the Basic Service Set wireless network, a simulation experiment can be carried out to assess the precise amount of calls that the G.729 codec can provide simultaneously and it’s respective MOS, delay, jitter and packet loss values. The effect of increasing the audio payload size form the default 20ms to 30ms can also be measured during the experiment.

3.0 Methodology

This section of the report presents the specific details of the primary research method. A number of methods are available however through the critical evaluation of literature; this section explains why the selected method was the most suitable way of testing the hypothesis.

3.1 Primary Research Method

An experimental approach will be used to test each hypothesis. In academic research, an experiments aim is to analyse cause and effect relationships (Oates 2005). The hypotheses which have been derived from the thorough literature review can now be tested empirically via an experiment. The experiment for this study will investigate the relationship between call quantity and call quality, therefore the explanation by Oates (2006) strongly relates to the experiment used for this study.

To conduct the experiment, a network simulation tool called OPNET Modeler will be used. OPNET provides an environment where the performance of computer systems, applications and communication networks can be specified, simulated and analysed. OPNET allows the parameters of networks, systems and applications to be specified in great detail whilst also boasting an extensive data collection facility which allows for accurate output of results.

Previous research has shown that experiments involving WLAN deployments have been carried out by Hneiti (2006), which allowed the hypotheses to be tested successfully. In addition, the study by Dao (2005) shows that a simulation approach can also be successful when analysing the relationship between call capacity and call quality. Both studies made use of the OPNET Modeler tool. Salah (2006) also used the OPNET Modeler tool with the aim of successfully deploying VoIP within a wireless network. Conclusions drawn from the above studies suggest that the simulation approach using OPNET Modeler is highly trusted, and that the results obtained from the experiment give a clear reflection on whether the hypotheses can be tested successfully or not.

The simulation approach also offers a great advantage when it comes to the flexibility of adding, removing or changing variables. Once the network topology has been created, the number of clients, voice codec used, and audio payload size can all be altered easily without having to interact with any physical pieces of equipment.

3.2 Alternative Methods

There are alternative methods available for assessing VoIP speech quality. Rather than use network simulation software, a real life experiment can be conducted using the necessary pieces of equipment.

One method of assessing speech quality in relation to call capacity is to physically build an 802.11g WLAN topology and generate the calls using call generator software such as SipP or WinSIP and traffic could be analysed using tools such as Omnipeek.

Previous studies have been carried out successfully when evaluating speech quality over WLAN deployments (Passito et al. 2005, Carvalho et al. 2005), however it could be argued that the software used to generate the voice calls may not give a true reflection of a genuine VoIP call.

To combat this, real VoIP phones could be used however this could difficult to implement as having to manage the physical VoIP handsets would become difficult.

Interviews or questionnaires could be carried out in environments where an 802.11g WLAN deployment using VoIP already exists however this would be not suitable at all as no technical data can be obtained to present an accurate performance analysis of the voice codecs. There are also no studies to suggest that this method has been used for this type of experiment.

3.3 Precise Nature of Intended Experiment

The network simulation experiment had taken into account the factors which were derived from the literature review. Using these factors, it was possible to build an 802.11g WLAN topology in an infrastructure setup to simulate VoIP phone calls.

A wireless topology was constructed in OPNET which consisted of one wireless access point. The number of clients associating with the access point increased with each scenario. The access points and clients were all able to transmit data at the data rate supported by the 802.11g WLAN standard.

An office environment of the size 300x300 metres was chosen as the scenario however since the purpose of the experiment was to analyse call quality in relation to call quantity, the background traffic feature was disabled. All calls were also set to be made from a quiet room.

The wireless clients were configured to transmit data which closely replicated how voice data would be transmitted by a mobile VoIP handset. The clients transmitted voice traffic based on speech with moments of silence, thus closely replicating a real voice sample(www.voiptroubleshooter.com). The traffic was transmitted for 60 seconds and was repeated 5 times, making the duration of each scenario 5 minutes. This allowed for average readings to be obtained. Each scenario transmitted voice traffic containing 20ms of audio; however the procedure above was repeated after increasing the audio payload size to 30ms.

At the end of the experiment, values for MOS, Delay, Jitter and Packet loss were obtained.

The wireless topology chosen for the simulation is modelled on the BSS (Infrastructure) topology which was described in the literature review. The other alternative was to use an ad-hoc network where no access point would have been needed. This method was not used as it does not closely represent a typical 802.11g WLAN deployment within an office environment. Also, ad-hoc clients tend to be within close proximity to each other, which makes the use of VoIP unpractical.

Both G.711 and G.729 codecs were used for this experiment as they are the most common codecs in use today in WLAN deployments.

The call signalling protocols available were discussed during the literature review. For the experiment, SIP was chosen as the signalling protocol due to its popularity compared to H.323. Performance levels between these two call signalling protocols are negligible thus having little effect on the results obtained from the simulation (Wallace 2008).

No QoS was implemented for the simulation as there was no interference from any other traffic sources. The aim of the study was to analyse call quality in relation to call quantity however further work may possibly be carried out in the future to analyse performance with and without QoS.

Below is a table which outlines the plan used for the simulation.

Table 2 - Scenarios for Simulation

Scenario	Codec	Signalling Protocol	Audio Payload	No. of Calls
1	G.729	SIP	20ms	1
2	G.729	SIP	20ms	5
3	G.729	SIP	20ms	10
4	G.729	SIP	20ms	15
5	G.729	SIP	20ms	20
6	G.729	SIP	20ms	25
7	G.729	SIP	20ms	30
8	G.729	SIP	20ms	35
9	G.729	SIP	20ms	40
10	G.729	SIP	20ms	45
11	G.729	SIP	20ms	50
12	G.729	SIP	20ms	55
13	G.729	SIP	20ms	60
14	G.729	SIP	20ms	65
15	G.729	SIP	20ms	70
16	G.729	SIP	20ms	75

17	G.729	SIP	20ms	80
18	G.729	SIP	20ms	1
19	G.711	SIP	20ms	5
20	G.711	SIP	20ms	10
21	G.711	SIP	20ms	15
22	G.711	SIP	20ms	20
23	G.711	SIP	20ms	25
24	G.711	SIP	20ms	30
25	G.711	SIP	20ms	35
26	G.711	SIP	20ms	40
27	G.711	SIP	20ms	45
28	G.711	SIP	20ms	50
29	G.711	SIP	20ms	55
30	G.711	SIP	20ms	60
31	G.711	SIP	20ms	65
32	G.711	SIP	20ms	70
33	G.711	SIP	20ms	75
34	G.711	SIP	20ms	80

As can be seen above, there are 16 possible scenarios each for both codecs using 20ms of audio payload only. The first 16 consist of testing the G.729 codec and the next 16 consists of the G.711 codecs. The first scenario places 1 call on the WLAN and is repeated a further 15 times with the number of calls increasing in increments of 5 from the second scenario onwards. Once 80 calls have been reached, the experiment was repeated using the G.729 codec. The entire process was repeated using an increased audio payload size of 30ms.

The problem with the above scenario was that the number of maximum calls possible was not able to be pinpointed to the exact unit due to the increase in calls by increments of 5 therefore once the average MOS score for a given scenario dropped below the threshold of 3.5, then the same scenario was repeated again however calls were increased by an increment of 1. For example if MOS < 3.5 at 60 calls, and > 3.5 at 55 calls, then steps were taken to calculate readings from 56 to 59 calls. This allowed the exact number of maximum calls possible to be pinpointed.

Once all outcomes were obtained, graphs for each scenario were produced providing a visual indication of performance for each scenario.

3.3.1 Configuration

Refer to Appendix I for screen shots showing configurations for the environment, topology and clients.

3.3.2 Network Topology

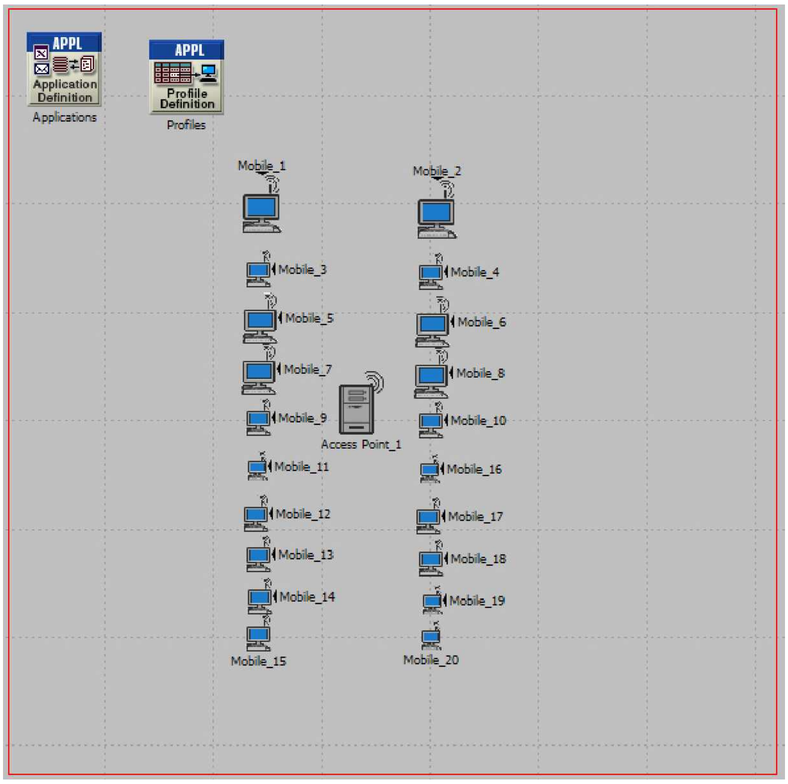


Figure 3 - WLAN Topology in OPNET Modeler

3.3.3 VoIP Profile Configuration

Refer to Appendix I for screen shots showing configuration for VoIP clients.

3.3.4 VoIP Application & Profile Configurations

Refer to Appendix I for screen shots showing configurations for VoIP Applications & Profiles

3.3.5 Resources

- Desktop PC running Windows XP Service Pack 2 and 4GB RAM
- OPNET Modeler version 14.5
- Microsoft Excel
- Microsoft Word

3.4 Validation of Setup

The network topology and configuration need to be validated before proceeding further with the experiment to avoid collecting invalid results.

The topology can be validated by using the "check consistency" feature within OPNET Modeler.

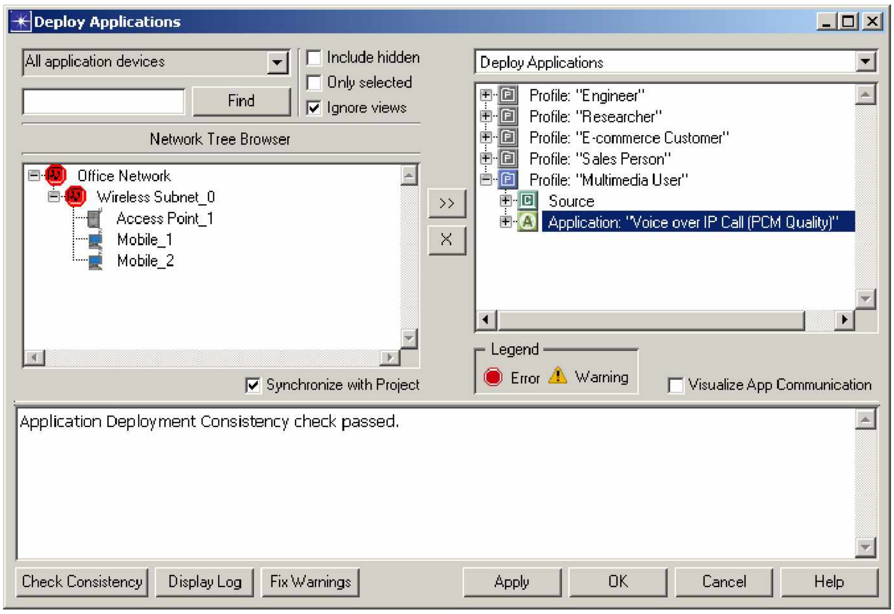


Figure 4 - OPNET Modeler Passed Consistency Check

Figure 4 above shows that the consistency check has passed.

The traffic generation of the configured VoIP application can also be validated by running a single telephone call between two clients to ensure that values for packet loss, delay, jitter and MOS are being output.

3.5 Procedure

This section describes the procedure involved in configuring and simulating a VoIP deployment on an 802.11g WLAN deployment in OPNET Modeler.

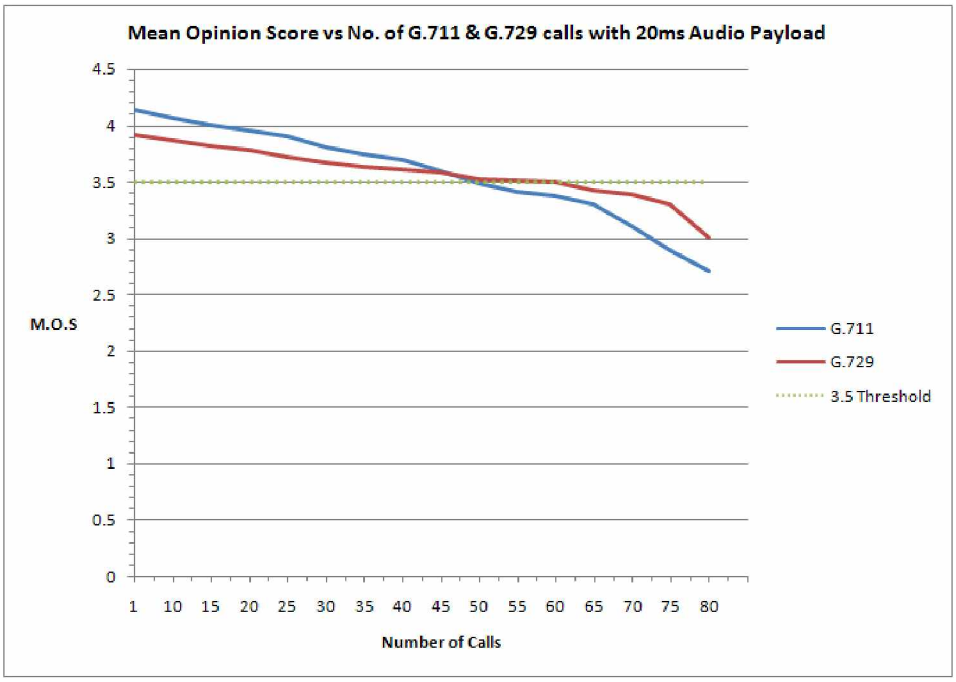
1. Run the Wireless network deployment wizard.
2. Configure a 300 x 300 metre office scenario. See Appendix I for screen shot of configuration page.

3. Select 802.11g WLAN network at the rate of 54mbps. See Appendix I for screen shot of configuration page.
4. Deploy one wireless access point with one wireless client
5. Configure VoIP application to use the G.711 codec.
6. Set Frame Size (Audio Payload) to 20ms and Coding Rate to 64Kbps. See Appendix I for further information.
7. Configure VoIP profile to use the application configured in step 6. See Appendix I for further information.
8. Deploy the VoIP profile configured in step 7 to the wireless client and enable the wireless access point to support the VoIP profile.
9. Duplicate the wireless client until two clients are available, therefore giving 1 voice call.
10. Set simulation to run for 5 minutes, and then run simulation.
11. Export results for delay, jitter, packet loss and MOS into Microsoft Excel and then calculate average values.
12. Repeat step 9 and increase number of calls to 5, and then increment the number of calls by 5 until 80 calls are reached.
13. When the MOS decreases below the threshold of 3.5 then the exact number of maximum calls has to be found. For example if MOS < 3.5 at 60 calls and > 3.5 at 55 calls, then the above steps are repeated to perform 56 – 59 calls.
14. Once the above is complete for the G.711 codec, repeat step 5 and configure the G.729 codec. See Appendix I for further information.
15. Once the necessary readings are gathered, repeat the above, however in step 6, increase the audio payload to 30ms.

4.0 Results

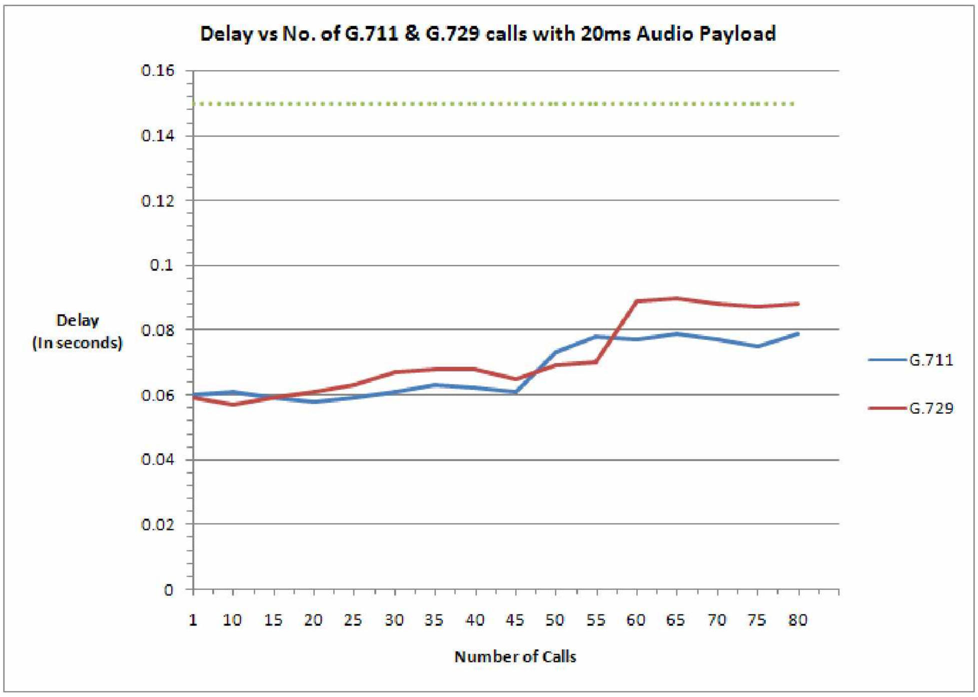
This section of the report displays the results and analysis of the simulation carried out as described in chapter 3.0. The results will analyse the performance of the G.711 and G.729 codecs and will identify which codec permits the maximum number of simultaneous voice calls whilst retaining acceptable speech quality. The effects of increasing the audio payload size from 20ms to 30ms are also analysed within this section of the report.

Graph 1 beneath shows how the speech quality of a call decreases as the number of calls increases. The G.729 codec which consumes less bandwidth than the G.711 codec is able to provide a higher number of simultaneous calls whilst retaining acceptable speech quality. For the G.729 codec, the MOS dropped below 3.5 at between 60 and 65 calls whilst the G.711 codec managed to retain acceptable speech quality until between 45 and 50 calls.



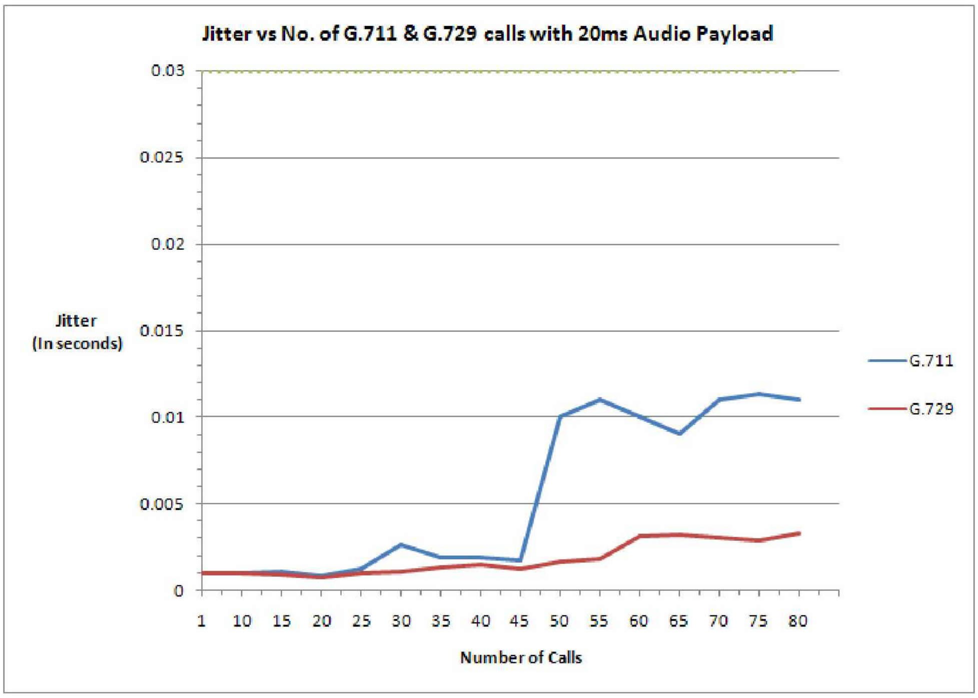
Graph 1 - MOS vs No. of Simultaneous Calls with 20ms Payload

Graph 2 below shows that delay did not cause any problems during the simulation as delay for both codecs never exceeded the 150 ms threshold.



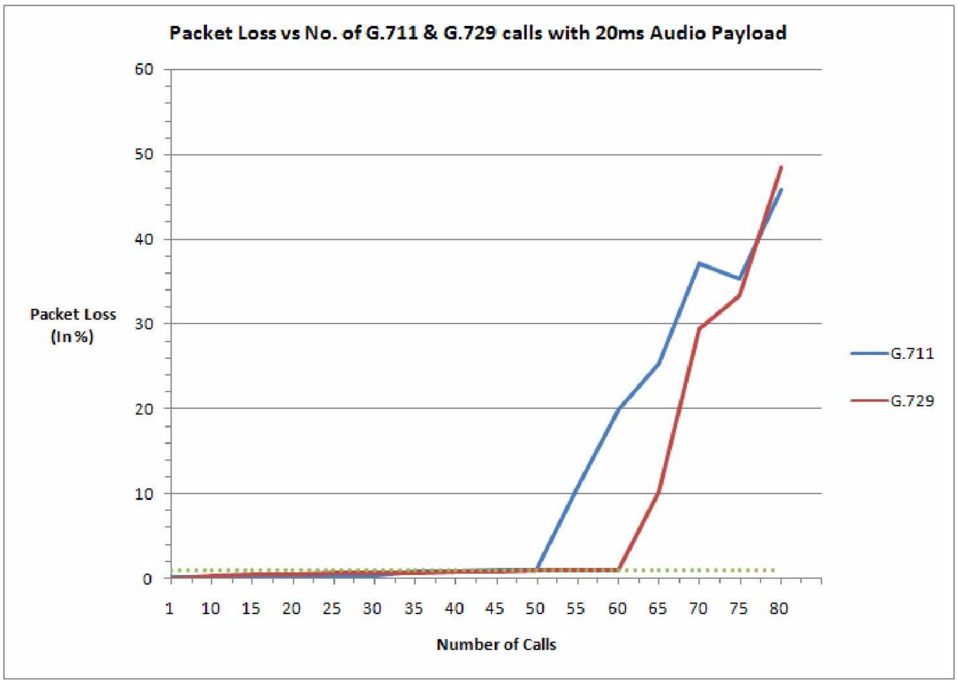
Graph 2 - Delay vs No. of Simultaneous Calls with 20ms Payload

Graph 3 below shows that jitter did not exceed the 30ms threshold and therefore did not cause any problems. The G.711 codec shows slightly higher jitter than the G.729 codec.



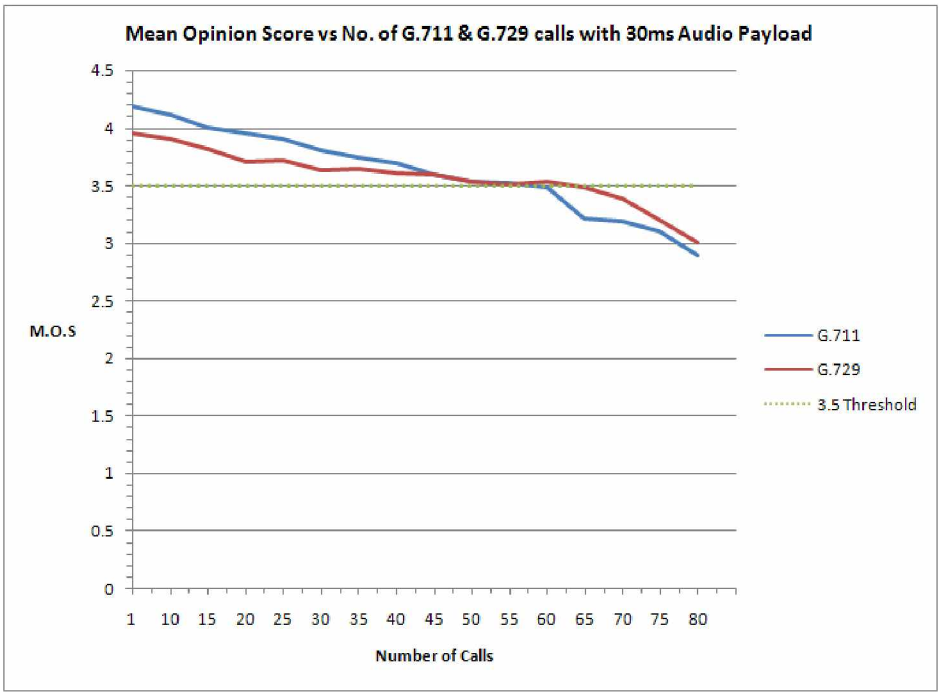
Graph 3 - Jitter vs No. of Simultaneous Calls with 20ms Payload

Graph 4 below shows that increase in packet loss can be directly related to the decrease of the MOS. Packet loss for both codecs was never greater than tha 1% threshold until the respective MOS values dropped below 3.5. Packet loss of up to 45% can occur once the number of calls reaches 75.



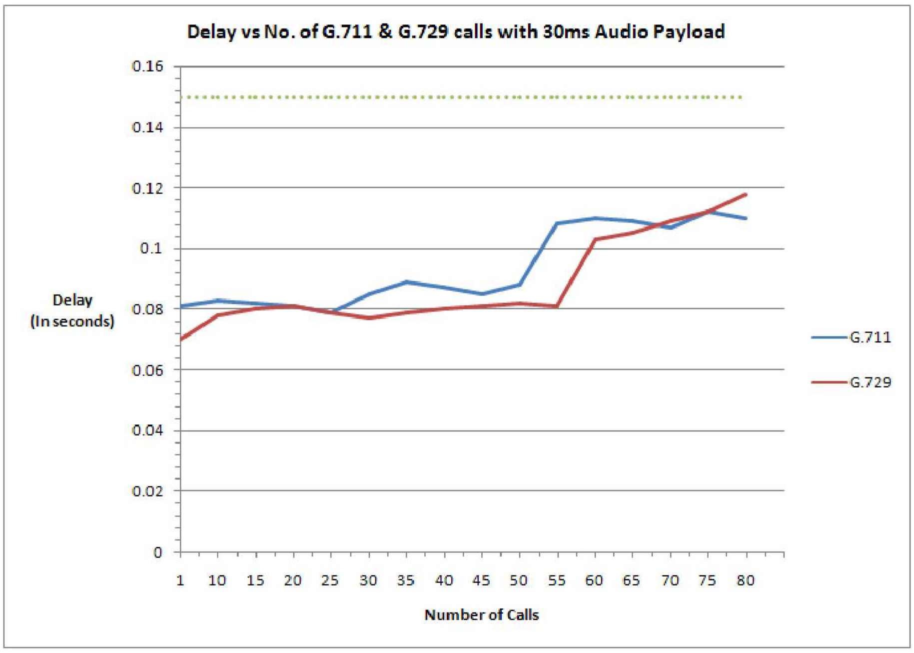
Graph 4 - Packet Loss vs No. of Simultaneous Calls with 20ms Payload

Graph 5 shows both codecs with increased audio payloads of 30ms. The results show that there is a slight increase in the maximum number of simultaneous calls possible whilst retaining call quality.



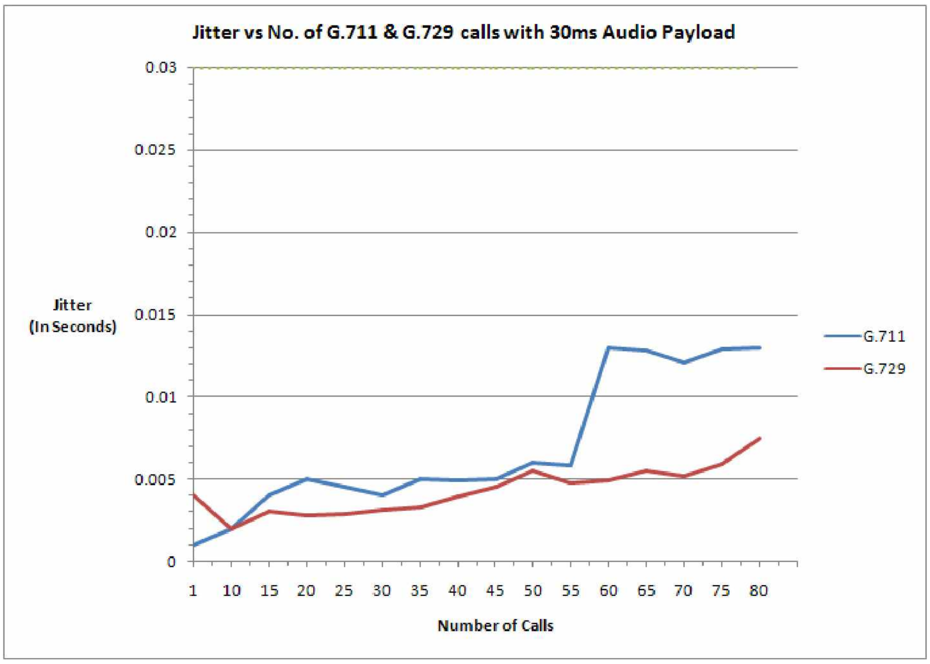
Graph 5- MOS vs No. of Simultaneous Calls with 30ms Payload

There is a slight increase in delay for both codecs and a slight rise at the point where MOS drops below 3.5 however delay does not exceed the threshold as shown in graph 6 below.



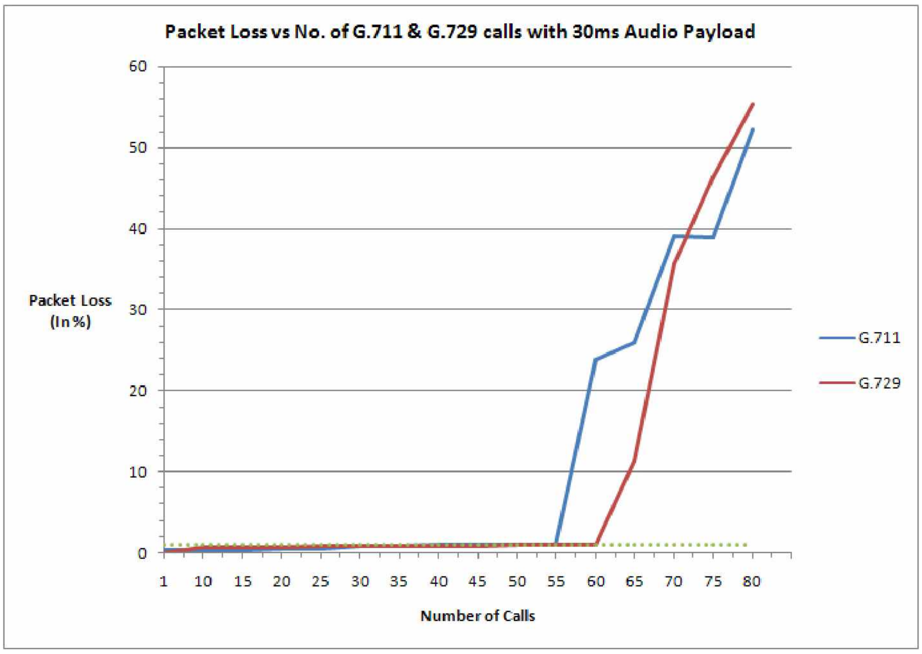
Graph 6 - Delay vs No. of Simultaneous Calls with 30ms Payload

Again, there is a slight increase in jitter for both codecs and a slight rise for the G.711 codec at the point where the MOS drops below 3.5 however jitter did not affect the simulation as it did not exceed the 30ms threshold.



Graph 7 - Jitter vs No. of Simultaneous Calls with 30ms Payload

Graph 8 below shows that packet loss is again directly related to MOS when using a 30ms payload. Packet loss does not exceed the 1% threshold until the MOS drops below 3.5. The packet loss rise is steep and can exceed 50% when performing 80 voice calls.



Graph 8 - Packet Loss vs No. of Simultaneous Calls with 30ms Payload

4.1 Evaluation

The maximum number of simultaneous calls possible in an IEEE 802.11g WLAN deployment for each codec is presented below in the table.

Table 3 - Summary of Maximum Number of Calls Possible

VoIP Codec	Maximum Number of Calls with 20ms Payload	Maximum Number of Calls with 30ms Payload	Percentage Difference
G.729	60	63	5% increase
G.711	48	55	14.6% increase

It can be seen from the table above that the G.729 codec allows the highest number of simultaneous calls possible within a WLAN deployment for both audio payload settings of 20ms and 30ms.

The G.729 codec allows 60 maximum calls with a 20ms audio payload before the MOS value drops below the threshold value of 3.5 whereas the G.711 codec allows 48 calls with a 20ms audio payload. By increasing the audio payload to 30ms, a slight increase in the maximum number of calls can be seen. G.729 permits 63 calls to be made before speech quality falls below 3.5 whilst G.711 allows 55 calls possible.

Although it is a slight increase, results show that the G.711 codec increased the most after increasing the audio payload size. G.711 allowed 14.6% more calls possible whilst the G.729 codec showed a 5% increase.

Results show that the increase in packet loss can be directly related to the decrease in speech quality. For both 20ms and 30ms audio payload sizes, packet loss did not exceed the 1% threshold until the MOS dropped below the 3.5 threshold. After exceeding the 1% threshold, the value shows a steep rise with packet loss showing above 45% for both codecs for 20ms payload, and above 50% for both codecs using 30ms of audio payload.

As can be seen in graphs 2, 3, 6 and 7, delay and jitter do not exceed their respective thresholds thus providing no problems during simulation. Graph 2 shows a slight rise in delay for both codecs using 20ms audio payload when the MOS drops below 3.5 however the delay is not sufficient enough for it to pose any issues. Graph 3 shows that the G.711 codec using 20ms audio payload experiences a slight rise in jitter by approximately 8ms when the MOS drops below 3.5 however it does not exceed the 30ms threshold. The G.729 codec from the same graph remains fairly level throughout experiencing slight rises and dips throughout. Starting from a slightly higher value in graphs 6 and 7, delay and jitter follow similar trends as they did when using 20ms of audio payload however they still do not exceed their respective thresholds and thus do not cause any issues during simulation.

5.0 Final Discussion & Conclusions

The overall objective of the project was to determine which VoIP codec would allow the most number of simultaneous 2-way phone calls in an IEEE 802.11g WLAN deployment whilst retaining acceptable speech quality with a Mean Opinion Score of 3.5 or above. The secondary objective was to analyse the effects of increasing the audio payload size.

The results show that the first hypothesis outlined in section 1.2 is true. The hypothesis states that the G.729 voice codec would allow the highest call capacity whilst retaining call quality. The G.729 codec uses compression to reduce the size of the packets being transmitted therefore utilising less bandwidth than the G.711 codec, which does not perform any compression. Although the initial MOS for the G.729 codec is lower than that of the G.711 codec, it does not fall below the 3.5 threshold before the MOS of the G.711 codec does therefore the maximum number of simultaneous calls on an 802.11g WLAN deployment is achieved by the G.729 codec. These findings are supported by previous work investigating the performance of G.711 and G.729 codecs and investigating call capacity in WLAN deployments. A study by Hole & Tobagi (2004) found that the G.729 voice codec is most suitable for use over WLAN due to its efficient bandwidth utilisation.

The second hypothesis outlined in section 1.2 was that by increasing the audio payload size from 20ms to 30ms, the maximum possible number of 2-way voice calls would increase. Results show that this hypothesis is also true. By increasing the audio payload size of a voice codec, the packet takes longer to construct however the packet contains 10ms extra audio therefore to transmit the same audio signal using 30ms of audio payload rather than 20ms, less packets are transmitted thus less bandwidth is utilised. The literature review also outlined that by increasing the audio payload, there is also an extra delay and higher degradation of speech quality in relation to packet loss. Although marginal, results show this to be true with the findings also backed up by previous research. The studies found by increasing audio payload size resulted in an increase in call capacity as the number of packets transmitted is reduced whilst also adding marginal overall delay (Hole, Tobagi 2004, Medepalli et al. 2004).

The values obtained for delay and jitter did not have a great affect on speech quality. Although there was a marginal increase in delay and jitter when increasing the audio payload from 20ms to 30ms, it was not enough for it to have a great impact on the MOS value. Previous studies have found that the delay does not have a great impact on the MOS value, even when delay exceed the 150ms threshold (Hole, Tobagi 2004).

A decrease in the MOS value showed a direct increase in packet loss. The 1% threshold was exceeded when the respective MOS value dropped below 3.5 for both codecs and for both audio payload sizes of 20ms and 30ms. This trend is backed up by Garg & Kappes (2003) where VoIP performance in an 802.11b network was analysed.

5.1 Project Critique

This section critically analyses the report and also discusses the strengths and weakness of the project itself. Alternative methods on how the project could have been improved are also discussed alongside any problems which arose during the development of the project.

The aims and objectives set out in section 1.2 of this report were all met through an extensive literature review. The first aim was to identify the common VoIP codecs available and how their technical attributes differ from one another. Completing this objective made it possible to identify which codecs could be used in the simulation. The second objective was to identify the various wireless topologies available. After a thorough literature review, this objective was met and identified the wireless topology which the simulation would be modelled upon. The third objective of the report was to identify the values which can affect speech quality, and to identify acceptable threshold values. Completing these three objectives provided the building blocks for the simulation to be carried out and to then obtain the necessary readings.

5.1.1 Project Strengths

This project has provided an insight into the two most common VoIP codecs available and how call quality is affected in relation to call quantity when using either of the codecs. The objectives set out in section 1.2 of the report have all been met by performing a thorough literature review which has provided the building blocks for a well structured project.

There is enough evidence to suggest that the findings from this report are robust and reliable as not only were the hypothesis in section 1.2 true, but the results obtained from the network simulation were also backed up by previous research.

5.1.2 Project Weaknesses

There are a few issues on how the project may have been improved. The network simulation tries to closely replicate a WLAN deployment within an office environment. Security is a major factor when it comes to WLAN deployment however there was no security implemented on the network topology constructed within OPNET. The reason for this was due to the fact that OPNET does not support WLAN security therefore even basic WPA2 security could not be implemented to secure the voice traffic. Previous studies have shown that the implementation of basic wireless security adds extra overheads to transmitted traffic therefore having a negative effect on WLAN performance(Bohn et al. 2006). The results of this project show the quality of speech relative to call quantity with no security overhead attached.

There is also the issue that the simulation was set to have zero background traffic. Since the aim of the project was to assess speech quality when using different codecs and varying amounts of calls, the background traffic option was disabled, however the chances of there being zero background ‘noise’ within an office environment are slim. The results drawn show speech quality with no influence from background traffic however previous studies has shown that background traffic can cause a decrease in voice performance(Ong, Khan 2008).

Some may argue that the OPNET Modeler simulation software may not provide a true reflection of a genuine VoIP call, and may not genuinely replicate real life performance of a WLAN deployment as the software is only as good as how it's designed and programmed however all scenarios were run in OPNET Modeler and their relative performance was analysed.

Also, the MOS values which were obtained were average values from each scenario therefore it could be possible that one call may have dropped below the 3.5 threshold however the overall average value could still be equal to or greater than 3.5. It could be argued that the maximum number of calls possible should stop when the MOS drops below the 3.5 threshold for the first time.

5.2 Further Work

There are various avenues which could be explored to build on the findings of this project.

5.2.1 Wireless Security

Wireless security is a must when it comes to implementing wireless networks to not only avoid data being intercepted but to also avoid voice traffic being eavesdropped. The implementation of WPA2 security can secure data and voice traffic but would also have a negative effect on network performance. The performance of network traffic being secured by other technology such as EAP-FAST and EAP-TLS could also be analysed. The use if IPSec to encrypt traffic could also be analysed. Research shows that implementing IPSec can have a crippling affect on bandwidth utilisation(Barbieri, Bruschi & Rosti 2002).

5.2.2 Voice Activity Detection

The Voice Activity Detection (VAD) technology is able to detect silences in speech on VoIP phone calls. Without VAD enabled, voice traffic is constantly being streamed across the network whether an individual is talking or not however with VAD enabled, it is able to detect silences within voice calls thus only streaming the data which has speech included. This can greatly reduce voice traffic by up to 35% which in turn reduces network congestion and increases voice quality (Haq et al. 2007).

5.2.3 802.11n

Although the 802.11n standard is not ratified yet, it has the potential to dramatically improve wireless network performance within office environments. The 802.11n standard provides up to 300mbps, with actual throughput being around half that at

150mbps; which is approximately 6 times more than what the actual throughput of 802.11g offers. The considerable amount of extra bandwidth will enhance user experience as network congestion will dramatically decrease. The other advantage of the 'n' standard is that it provides greater network coverage resulting in less access points having to be deployed. The 'n' standard however will not make its way into corporate environments until it has been ratified.

5.2.4 Network Topologies

The performance of VoIP codecs could be analysed by implementing 2 access points where traffic would have to be transmitted over both access points. This would allow for greater wireless coverage and extra bandwidth available and would be ideally analysed using a network simulation tool.

5.2.5 Call Signalling Protocols

The simulation for this study used the SIP protocol however the performance for H.323 could also be investigated although research does show that there is minimal difference between the two protocols. Other signalling protocols could be tested such as Media Gateway Control Protocol (MGCP).

5.3 Conclusion

The performance of VoIP codecs in an IEEE 802.11g WLAN deployment was analysed. Results showed that the G.729 codec provided the highest number of simultaneous 2-way calls whilst retaining acceptable speech quality. The G.729 codec managed 60 simultaneous voice calls with a 20ms audio payload and a slight increase to 63 simultaneous calls with a 30ms audio payload. The G.711 codec managed 48 simultaneous calls when using 20ms of audio payload and increased to 55 simultaneous calls when using 30ms of audio payload. Results also showed that packet loss increased as a direct result of the decrease of speech quality. Increasing the size of the audio payload from 20ms to 30ms lead to a marginal increase of maximum number of calls with acceptable speech quality.

6.0 References

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7.0 Appendix I - OPNET Modeler Configuration

7.1 OPNET Modeler WLAN Deployment Wizard Setup pt 1

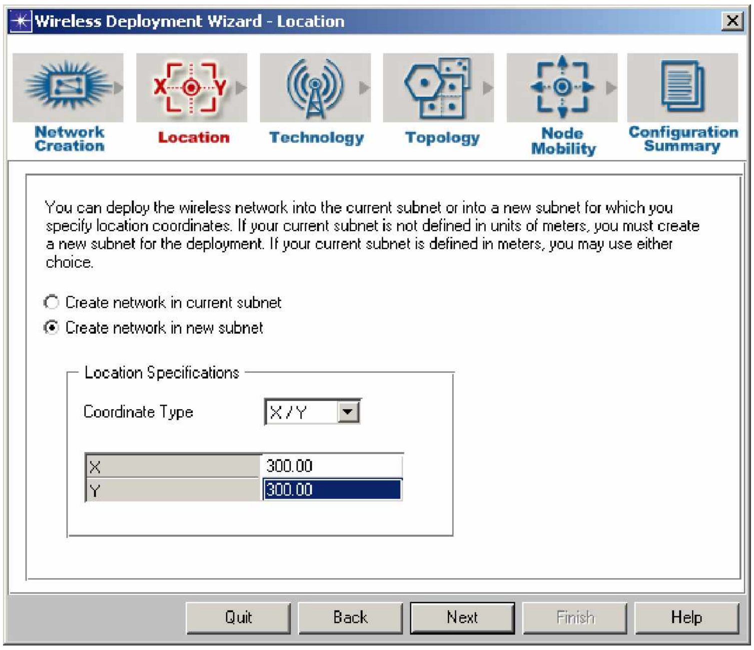

Figure 5 - OPNET Modeler Office Coordinates Setup

7.2 OPNET Modeler WLAN Deployment Wizard Setup pt 1

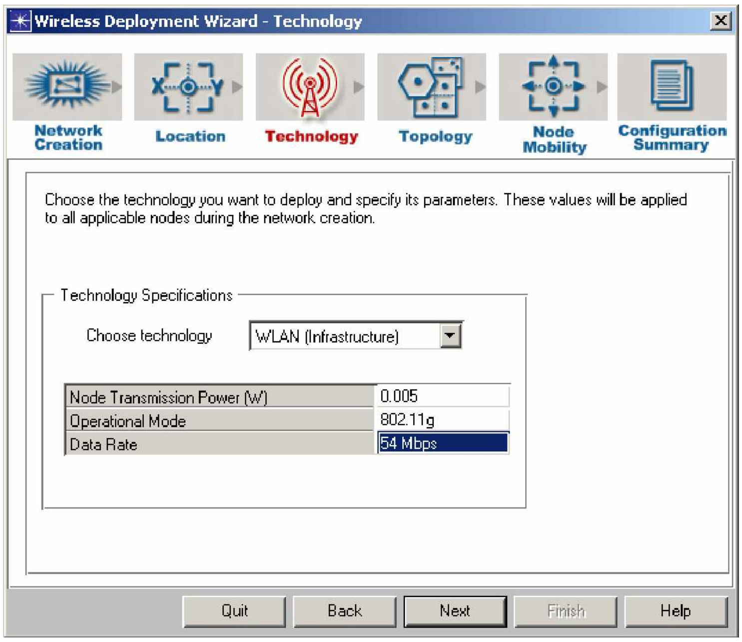

Figure 6 - OPNET Modeler WLAN Deployment Wizard

7.3 OPNET Modeler G.711 VoIP Application Setup

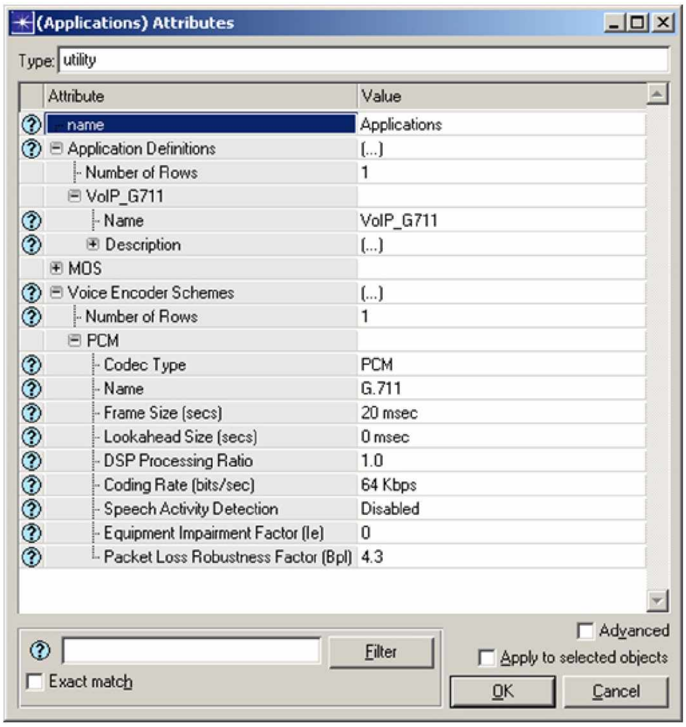

Figure 7 - G.711 VoIP Application Setup

7.4 OPNET Modeler G.729 VoIP Application Setup

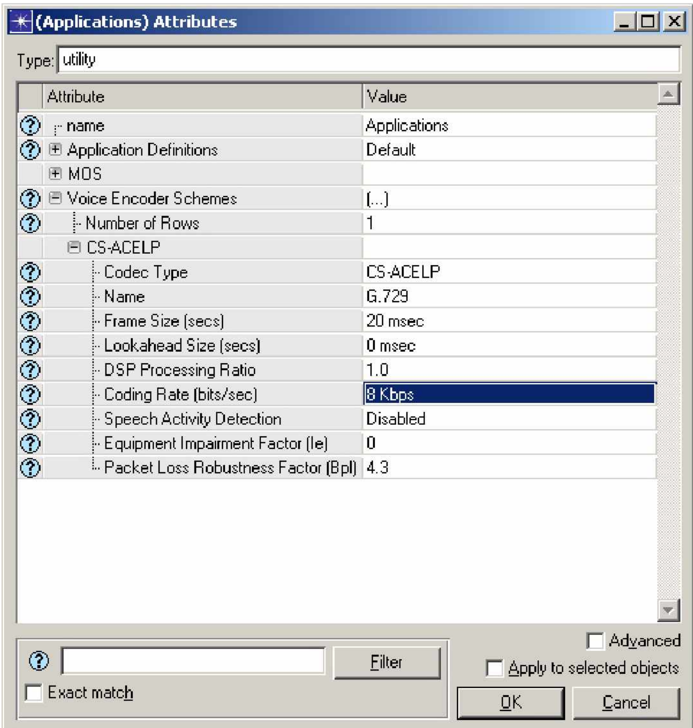

Figure 8 - G.729 VoIP Application Setup

7.5 OPNET Modeler VoIP Profile Setup

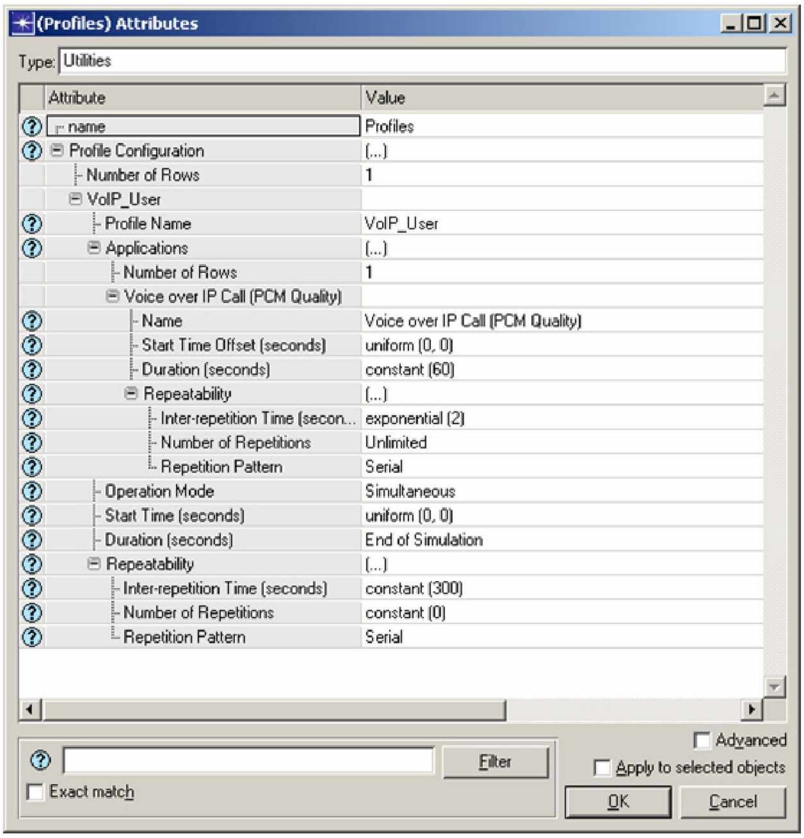

Figure 9 - VoIP Profile Configuration