Final Honours Report

"What is the call capacity of VoIP and video conferencing when delay, jitter and packet loss are the metrics of QoS for each end to end call when simulated over WAP, MANET & ESS wireless technology environments, which are populated with dynamically moving nodes."

By
Judge Doom
2005xxxxx

BSNE4/5_1

Project Supervisor: Dimitrios Liarokapis
2nd Marker: Richard Foley

"Except where explicitly stated all work in this document is my own."

Signed:________________________ Date_____________

Abstract

The increase in the Quality of Services (QoS) that wireless technologies are now able to provide has resulted in the integration of multimedia traffic, in particular voice (VoIP) and video conferencing onto the wireless medium. Wireless technologies in regards to VoIP and video conferencing traffic are pushing the boundaries of using wireless technologies further than simply accessing the Internet and data transfers. With increasing numbers of Wi-Fi enabled devices that allow VoIP and video conferencing applications it thus raises the question as to whether or not different wireless technologies will be able to cope with the increase in users' demands.

The aim of this report was to investigate through network simulation whether WAP, MANET and ESS wireless technologies are able to cope with the increasing demand through identifying the call capacity of VoIP and video conferencing over each wireless technology with the deployment of mobile nodes. Node movement within the various environments aimed to reflect as accurately as possible real life user movement, to achieve test data as realistic to that which would be obtained by physically carrying out the experiment. This being achieved through nodes being assigned one of three mobility profiles that ranged from low, medium and high movement distances relative to each environment size. With three different background flow demands, low, medium and high, along with the number of flow, 25%, 50%, 75% and 100% of the node population meant that each call could be tested under 12 different scenarios to identify how it impacts on the call performance. From the results collected and the ITU recommendations in regards to VoIP and video conferencing call quality led to the call capacity within each wireless technology environment being identified.

The findings of this report clearly demonstrate that video conferencing additional media stream of video results in an increase in packet size compared to VoIP; subsequently making it not possible to achieve a single video conferencing call over any of the wireless technologies. The reason for this is indicated by the high number of bits that are dropped due to the video buffer overflow because the WAP is unable to empty the buffer quicker than the number of new packets arriving. VoIP traffic with its smaller and more frequent packet characteristics compared to video conferencing led to the conclusion as to why VoIP calls that met the ITU recommendations were successful in nearly all of the tested wireless environments. It was only in the MANET environment due to its lack of infrastructure that it was observed that there was no consistency between runs or scenarios. This is demonstrated through the irregular call capacity identified between scenarios and is documented as the direct consequence of node movement effecting the performance of a MANET network.

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1. Project Introduction

The following section of this report is an introduction as to the growth of wireless technologies, and how the introduction of VoIP and video conferencing has not only pushed the boundaries but also the capabilities of wireless technologies.

1.1 Background

Wireless Technologies in the 21st Century

Wireless technologies are increasing becoming a fundamental part of everyday life, affecting individuals not only within the work environment but also within all other aspects of their life. It is anticipated that the wireless network industry will be valued at $19 billion in 2012, with an estimated 350 million people using Wi-Fi and new Wi-Fi hotspots will be added to the currently existing 200,000 that are already deployed (Racino-Farrer 2007). This highlights that wireless technologies are expected to continue to grow and become a ubiquitous part of our daily lives. With already half of America having once used wireless access emphasises the use of this technology and the impact it is having on people's lives (Clancy 2009). It is thus moving towards T.Kuang's vision of mobile network access allowing users "anything, anytime, anywhere" access due to the increasing wireless technology coverage (Kuang, Williamson 2005). This growth of technology can also be seen to be impacting on the younger generations with schools and universities investing heavily in this type of technology. The 2006 National Survey of Information Technology in the U.S. Higher Education report concluded that "three-fifths (60.5 percent) of colleges and universities increased their campus IT budgets for wireless for the current academic year" (Green 2006).

Demands over Wireless

Steve Jobs, Chief Executive Officer of Apple, envisioned wireless to be a "liberating experience to surf the Internet while freely moving about your home or classroom" (Jobs 1999). Since 1999 the boundaries and possible uses of wireless technology have been expanding as a result of enterprises wanting to achieve more from this technology (Barr, Galperin 2004). By enterprises pushing the boundaries of wireless technology new standards for wireless technologies have had to be developed to address this. The Institute of Electrical and Electronics Engineers (IEEE) have published wireless standards such as 802.11b, 802.11g, 802.11e and 802.11n (IEEE 2009) and mobile ad hoc network (MANET) by the Internet and Engineering Task Force (IETF) to try and meet the changing demands. The boundaries that are currently being pushed are: increase of multimedia applications and user mobility which can be linked to a 2008 report by Abyzov et al which identifies that "new devices lead to changes in the way that WLANS are used." These new devices and their functionality have lead to users requiring continuous wireless connectivity whilst they move through environments. At the ACM 'SIGCOMM' conference in August 2001 after monitoring the wireless network performance over the three day conference their findings concluded that greater than 80% of the users on the first two days were seen at more than one Wireless Access Point (WAP) (Balachandran et al. 2002). With increasing numbers of user devices being sold and the increasing use of multimedia applications in particular Voice over IP (VoIP) and video conferencing means that wireless networks are encountering new demands, thus pushing the boundaries of wireless technologies even further.

VoIP

VoIP allows users to make phone calls over an IP network. It is widely viewed to have several benefits over that of a traditional phone system including cost saving, improved call quality and integrated data and voice network (Cisco Systems Inc. 2004b). Early deployment of VoIP within enterprise environments were not successful and resulted in many companies either not implementing it or after trying it reverting back to their previous telephone solution. This is indicated in Appendix 1 which shows that approximately 70% of people said the reason for not implementing a VoIP solution was due to poor call quality. The nature of voice calls makes it very sensitive to the effects of delay and interruption in transmission. A significant factor was the result of Quality of Service (QoS) required in handling VoIP not being in place to provide a service which the users expect to be "nothing less than the reliability and accessibility of today's telephone network for their everyday communication." (Ried 1999). As technology has improved and network infrastructure upgraded it has resulted in VoIP being redeployed into enterprises with it being expected by 2010 that 95% of big companies will have started to converge their network to include VoIP (Dudman 2006). This is a result of the upgrades being able to provide a better QoS for VoIP, as indicated in Appendix 2 by the increase from 2007 to 2009 of user ratings in relation to call quality, showing that VoIP is now on par or better than traditional analogue telephones.

With mobile devices, in particular mobile phones, providing VoIP functionality it has resulted in users and enterprises using potential free wireless access points to call other VoIP users. Adam Boone, Vice President at Sipera, is aware of a client that utilizes this functionality logging more than 1 million minutes per month in VoIP calls as a direct impact of the financial savings through not requiring the service of carrier networks (Goodin 2009). Skype which is a free VoIP software application is increasingly becoming available on smart phones indicating that future VoIP will be further utilized through these mobile devices and accessed using wireless technology. It was reported that on 5th October 2009 there were 19,142,581 concurrent Skype users online and this was predicted to increase to 20 million by November 2009 (Barton 2009), thus highlighting the fact that there is a demand for VoIP.

Video Conferencing

Video conferencing has also increased in popularity in the same way that VoIP has. The additional service video conferencing provides over that of VoIP is its real time stream of video allowing the user to both see and hear the user at the other end of the call. A 2005 (Walsh) online press release from Microsoft highlighted the popularity of its MSN messenger video conferencing solution with it being used by 30 million of its 155 million users each month.

Video conferencing enriches the experience of the communication between two or more users by providing a visual feed between the caller and called party. It has started to be used by more enterprises due to the benefits it can offer. For example:

- **Carbon Foot Print**: No additional travelling is required.
- **Financial**: No additional costs in relation to travel and accommodation.
- **Time**: No time being lost due to having to travel to meetings. Also meetings can be arranged at short notice, as no travelling time has to be factored in.

World events and disasters, such as 9/11, can have a dramatic impact on the investment in video conferencing technology. It was reported that BT saw a "rise by 85% in the week following the attacks" on the World Trade Centres (Edser 2001). One video conferencing company, Regus, experienced a 230% increase in the use of their 2500 video conference suites following the Icelandic volcano, Eyjafjallajökull, erupting in April 2010, due to air travel being grounded, predominately in Europe, for a significant period of time. What was of significance was the Regus chief executive identified that the increased demand was from new customers, stating it has “moved video communication to the very front of business people’s minds as a cost-effective alternative to international business travel.” (Tobin 2010).

This increase does not support Ochsman and Chapani 1974 research which concluded that “there is no evidence in this study that the addition of a video channel has any significant effects on communication times or on communication behaviour” when compared to telephone calls (Chapanis, Ochsman 1974). However in a more recent study on video conferencing within an enterprise, users stated “having video improved the communication among the team” (Tang 1992). The contrast could be the result that users are interacting with multimedia more in their daily life in the 1990s compared to 1970s. Mobile users that have conference call capabilities on their phone are now benefiting from the increasing number of Wi-Fi Hotspots that are available not only for business but also for personal use. Global financial difficulties and increased terror attacks mean that video conferencing solutions in the future will become more common business practice for an enterprise of any scale. The trend that ‘social networking’ sites pursue in the future will also affect the social use of video conferencing functionality since Facebook alone has 350 million active users of which 65 million access it from mobile devices (Facebook 2010).

Video conferencing like VoIP is pushing the boundaries of wireless technologies and as a result the level of QoS provided to the user can vary due to different factors which will be discussed in the next section.

The Boundaries that VoIP and Video Conferencing are Pushing

As stated earlier wireless boundaries are being pushed by enterprises wanting to achieve more over the wireless media. Wireless was not originally designed to handle such time sensitive traffic such as VoIP and video conferencing. It works on the principle of Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), it therefore only allows one device to send or receive data over the medium at anyone time (Cisco Systems Inc. 2004b). This adds to the nature of wireless signals being sent through the air making it highly vulnerable to interference which can further reduce the performance and the QoS users experience. Delay, latency, jitter and packet loss are commonly used performance metrics that are used to identify the QoS (IEEE 2009). If the metrics’ values are high then the user may experience jerky video movement or gaps in speech flow resulting in the user having to end the call.

The new demand on wireless networks affects the three widely deployed application-layer network communications methods: Wireless Access Point (WAP), Mobile Ad Hoc Network (MANET) and Extended Service Set (ESS). Despite each technology using wireless as its sending media it is not possible to perform a direct comparison due to their characteristics and implementation being aimed at different environments and scenarios. These characteristic changes mean that the performance of VoIP and video conferencing performance over each technology will potentially vary.

Why Evaluating VoIP & Video Conferencing over Wireless Technologies is Important?

The inevitable increase in wireless technology means that VoIP and video conferencing will continue to progress onto a wireless medium, with enterprises converging their network infrastructure onto wireless networks due to its fast and cable free deployment potential. With 491 million expected cellular handsets to contain a Wi-Fi chipset globally by 2012 the number of users capable of accessing Wi-Fi hotspots suggests that the demand for wireless will increase (Clancy 2009). With VoIP and video conferencing offering potential cost benefits, increased functionality and improved quality compared to existing services means that more people will be able to use them. This adoption by more users means that wireless access points will become more populated by users, subsequently putting more demand on the different wireless technologies. This has resulted in the boundaries of wireless being continually pushed to try and reflect the demands capable of that on a wired Ethernet network referred to in Bar and Galperin 2004 report as ‘cordless Ethernet’. Ultimately when the boundaries of wireless technology have been reached and cannot be pushed any further then the whole wireless technology will have to be revised to address these new demands.

The aim of this project is therefore to identify the boundaries of the different wireless technologies specifically in regards to the call capacity of VoIP and video conferencing of mobile nodes. Through identifying this it will help the wireless industry and enterprises to support previous research or to identify if it is a feasible solution for specific enterprises.

1.2 Project Outline & Research Question

The outline of this project is to investigate what the call capacity of VoIP and video conferencing is over wireless technologies WAP, MANET and ESS using mobile nodes. This is to reflect the expected increase in mobile devices in the future that will be able to utilize wireless technologies and implement these types of calls. By changing different characteristics within each environment, for example the number of calls and background traffic, it will provide information as to how each characteristic affects call capacity and QoS traffic handling. This research will provide further information on what the current wireless technology boundaries are in regards to VoIP and video conferencing, subsequently identifying whether the current technology is capable of meeting expected user demands in the future.

From this, the research question of this project has been derived and is stated below:

> “What is the call capacity of VoIP and video conferencing when delay, jitter and packet loss are the metrics of QoS for each end to end call when simulated over WAP, MANET & ESS wireless technology environments, which are populated with dynamically moving nodes.”

1.2.1 Project Aim

The project aims to highlight the boundaries that wireless technologies are currently being pushed towards. As previously discussed VoIP and video conferencing are trying to be used over wireless technology and this is expected to continue into the future. The outcome of this project aims to identify what the call capacity is when made over different wireless technologies.

To achieve this, a number of objectives have to be addressed and met, and these are noted below.

Number 1: Create three wireless environments that reflect ‘real world’ environments in relation to each wireless technology.

Object purpose
To ensure the results are as accurate as possible the correct environment has to be designed to reflect the environment that the wireless technology would be implemented in within the ‘real world’. If this is not achieved then the data gathered will not be as credible and have a direct impact on the reliability of the conclusions of the project.

Number 2: Identify user movement within a ‘real world’ environment.

Object purpose
To identify how nodes move in the environment to increase the credibility of the results through accurate node movement within the test simulations. The speed of the node is just one factor of this and particularly within MANET environment where it may have a significant impact on the test data gathered.

Number 3: Identify call capacity of VoIP and video conferencing calls over each wireless technology.

Object purpose
This will identify when the network is unable to handle anymore calls due to the wireless technology limitations being reached.

Number 4: Analyse the quality of the calls to identify whether they meet the ITU recommendations.

Object purpose
This will identify when the quality of the calls fall below the ITU recommendations, subsequently identifying what the call limitations are for each specific environment. When calls fall below the recommendations it is deemed that the quality of the call is of a level that is unacceptable for the user to continue to use.

Number 5: What effect does changing the characteristics of the environment have on the call capacity?

Object purpose
From structured testing the call capacity will be analysed to identify when characteristics of the environment are changed and how it impacts on the call capacity. Such changes include number of background flows and the type of background traffic.

Number 6: Identify the call capacity to node population ratio in each environment.

Object purpose
This will try to identify if the call capacity scales to the node population in each environment. ESS and MANET should provide the most interesting results because of ESS multiple WAPs topology and MANET nodes ability to route packets.

Number 7: Analyse the effect of the additional media stream of video used in video conferencing and how it affects the call capacity in relation to VoIP calls.

Object purpose
This will identify how the severity of adding an additional media stream (video) affects the call capacity in each environment.

1.3 Hypotheses

Below are the three hypotheses which have been derived from the literature review in section 2.

1. In a WAP environment the call capacity will be VoIP close to 7-8 calls, with video conferencing being less.

Justification
From the literature research carried out it identifies from a range of creditable sources that the maximum call capacity of using a single WAP implementing 802.11b standard is 7-8 calls. The most credible source is from the Cisco book 'Deploying Voice over Wireless LANs' which states for 802.11b technology is able to support 8.54 (Geier 2007) and in a report by the computer chip manufacturer 'Intel' maximum of 7 calls is possible (Bakre 2006). We can expect slightly higher call capacity due to the implementation of 802.11g standard that provides higher bit rate of 54Mbits/s compared to 802.11b 11Mbit/s. The introduction of background traffic into the network should not affect the maximum capacity of VoIP calls due to the 802.11e priority queue algorithm and Davis and Keegan (2006) research identifying traffic up to 256 Bytes packets has little effect on call quality thus affecting call capacity. We predict lower call capacity in regards to video conferencing due to the 18240 bits per second per call increase that has to be sent due to the additional media stream of video introduces.

2. In the MANET environment there will be a node capacity that the performance will decrease due to either there being too few or too many nodes.

Justification
Kleinrock and Silvester (1978) literature identifies that in a MANET environment the optimum number of neighbouring nodes a node should have is six. A more recent study carried out in 2001 by Royer et al however concludes that 7-8 nodes is the optimum number of neighbouring nodes. From these studies two results can be expected in the 200 node MANET environment. The first is due to a low node count, with the potential number of routes to destinations being less than that of the 400 node environment thus decreasing the probability of a route to the destination node being reached. The other extreme is that because of the lower node count the delivery of packets has fewer hops to the destination thus reducing the end-to-end transmission time.

Optimum performance is expected to be achieved when the environment has 400 nodes due to the number of nodes increasing the probability of a node having 7-8 neighbours through out the environment compared to 200 node environment. With 7-8 nodes achieving the optimum throughput it is expected peak call capacity will be reached under these conditions.

3. The total call capacity in ESS environment will not scale to: the number of AP's x calls capacity of a single WAP.

Justification
In ESS APs are distributed across medium to large environment to provide continuous coverage to users. It could be assumed installing more APs will increase the call capacity of the environment by the number of APs installed; however this is not predicted from the findings of the literature review. Thus is due to like the nodes, APs in the ESS infrastructure have to contend to access the medium, limiting the call capacity. It is foreseen that the simultaneous number of calls will be higher than WAP but due to node 'handoff' period potentially lasting up to 300ms (Cisco Systems Inc. 2004b) during which the node cannot send or receive data. It is a critical factor which is directly influenced by node movement within the environment and result in a call experiencing

delay higher than the recommended 150ms (Cisco Systems Inc. 2004b). Another factor is calls will be made that will go across multiple APs to reach the destination called party. Thus identified from El-Hennawey et al (2009) report that identified that over two hops the call capacity drops by 25% due to the call quality decreasing. Through the call's hop count increasing it ultimately leads to only a single call being supported over a five hop route.

1.4 Report Structure

The remainder of this report is structured as discussed below.

1.4.1 Literature Review

In section 2, the literature review, aims to provide an understanding of the technologies and characteristics that make up the simulation environments. A variety of books, user manuals, reports and conference proceedings are used to reference previous and current work in the related areas thus providing a balanced literature review.

1.4.2 Methods

The methods for this project is presented in section 3, which provides specific details on how the primary research will be carried out with appropriate justification for choosing such methods. The specific details such as chosen environment characteristics, node mobility speed and call configurations will be specified with the supported literature to validate the data being used to carry out the experiment. In addition to the set up of the simulation it will identify what data will be gathered during the simulation and why it is relevant to the objectives this project sets out to achieve.

1.4.3 Presentation of Results

The finding from the primary research, which in this report was achieved through network simulation are presented. Through the demonstration of a variety of statistics gathered the call capacity of each environment is identified and how it is affected when the environment characteristics are changed for example the number of background traffic flows. The reason why the call capacity has been reached is also discussed in this section through using additional statistics collected from that of the ITU recommendations.

1.4.4 Summary and Conclusions

From the completion of the report this section evaluates the results obtained and what has been learnt as a result of carrying out this project. The project mythology and hypothesis are critical evaluated to identify if this project has been a 'fair study' of the subject area and as a result what future research may be carried out as a result of the findings of this project.

2. Literature Review

The literature review, as documented in section 1.4.1 aims to provide an understanding and discussion of the technology and characteristics' key to this project. Through critical analysis of the questions below it will be possible to draw conclusions derived from a variety of sources.

The questions that the literature review aims to address are:

- Overview of Wireless Technology & identifying characteristics of WAP, MANET and ESS topologies.
- Analysis into the characteristics of VoIP and video conferencing traffic.
- Analysis of QoS in regards to supporting VoIP and video conferencing traffic.
- Analysis of call capacity of each topology
- Identify characteristics of node mobility and density

2.1 Overview of Wireless Technology & identifying characteristics of WAP, MANET and ESS topologies

This section will discuss how the wireless technology works and common problems when implementing this technology, further to this the characteristics of each topology will be discussed

2.1.1 Overview of Wireless Technology

Wireless technology (802.11) as discussed previous is an IEEE (Institute of Electrical and Electronics Engineers) standard first released in 1997 (IEEE 2009). The technology brought number of benefits over wired solutions including mobility and reduced installation time, together with both short and long term savings (Cisco Systems Inc. 2004b). However during first deployment of this technology it had a so called 'false start' for reasons including immature technology, lack of standards and slow throughout speed. (Cisco Systems Inc. 2004b).

These initial problems have been addressed as can be seen from section 1 by the increasing adoption of this technology. However despite overcoming these problems from the development in IEEE 802.11 'a', 'b', 'g' and 'n' standards, new demands in 2010 is increasing the result in multimedia traffic specifically in regards to VoIP and video conferencing as previously discussed.

2.1.2 Shared Characteristics

The implementations of wireless in different topologies of WAP, MANET and ESS have several common characteristics and problems that affect there performance:

Access method to the medium

To prevent collisions occurring when nodes try to send across the medium within a wireless environment a fundamental mechanism that operates on the MAC layer is called Carrier Sense Multiple Access Collision Avoidance (CSMA/CA). It is employed by an access method called Distributed Coordination Function (DCF), which can be implemented within any wireless environment unlike the other method called Point Coordination Function (PCF), which requires infrastructure backbone that WAP and ESS provides. Unlike Ethernet, which implements Carrier Sense Multiple Access Collision Detection (CSMA/CD) where a collision can be detected on the cable medium, wireless implements CSMA/CA, which prevents the possibility of a collision occurring due to it not being possible to detect collisions in midair. As a result only one node can send at a time therefore nodes including APs contend to access the medium. This access mechanism ultimately was not designed to handle time sensitive traffic such as VoIP and video conferencing, where delay has a significant impact on the quality of the call, thus leading to the development of 802.11e (Cisco Systems Inc. 2004b) and 802.11n (Geier 2007) to address this issue.

Hidden node problem

Hidden node problem, as illustrated in diagram 1 below, occurs when nodes are out with the range of other nodes transition range, thus they assume the medium is clear to send leading to a collision to occur. From the diagram nodes A and C assume that the medium is free because they are out of transmission range of each other thus they send data to node B causing a collision to occur. As node movement increases within an environment the probability of hidden node occurring increases. In a dynamic environment that of an Ad Hoc topology the hidden node problem could occur more frequently and simultaneously throughout the environment due to its lack of infrastructure.

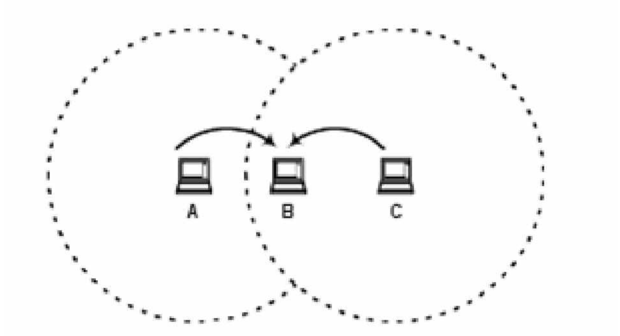


Diagram 1. The Hidden node problem (Anastasi et al 2004)

To address this problem there are several proposed solutions from (allexperts 2010) and (Laurenson, Tsertou 2006). The first method extends the DCF mechanism by using two control frames, Request to Send and Request to Clear (RTS\CTS). This is when the source node sends RTS frames to the destination before data frames, with the destination replying with a CTS frame identifying that the source is able to send. Another method proposed is token passing, in which a host can only send data if the node is in possession of the token. These mechanisms whilst addressing this problem do have an impact on performance through lower throughput. Anastasi's 2004 publication identified that in some cases a drop of 80KBps throughput in performance when RTS\CTS was implemented. This drop in level of performance could affect VoIP and video conferencing because of the lower throughput rates so a decision of probability of hidden node occurring has to be offset by the decrease in performance through implementing a solution.

Obstacles

The nature of the wireless medium being that of radio waves mean that the environment and the obstacles within it will have varying levels of impact on the performance that the nodes will experience whilst roaming within the environment. Such objects that impact on node performance include cordless phones, microwaves, trees and the building infrastructure itself like wall thickness and material. This is highlighted in Chatzigiannakis et al (2006) study into how various shaped objects affected performance and it concluded that the presence of obstacles do in fact have a negative impact on performance. Diagram 2, below, shows how radio waves can bounce off obstacles and as a result take significantly different paths depending on the object characteristics thus varying performance levels. In the case of WAP and ESS deployment fixed objects can be either removed or additional wireless devices such as repeaters or APs can be installed. This reduces the impact that obstacles have on performance of the nodes in the environment. In an Ad Hoc environment this is not possible due to its infrastructureless architecture, this resulting in it not being possible to accommodate through additional devices using this wireless topology. Other obstacles that are not as noticeable or visible but still have an impact on performance in an outdoor environment include rain, wind and fog as documented in the Braun et al 2001 publication called "High Performance Wireless Networking and Weather".

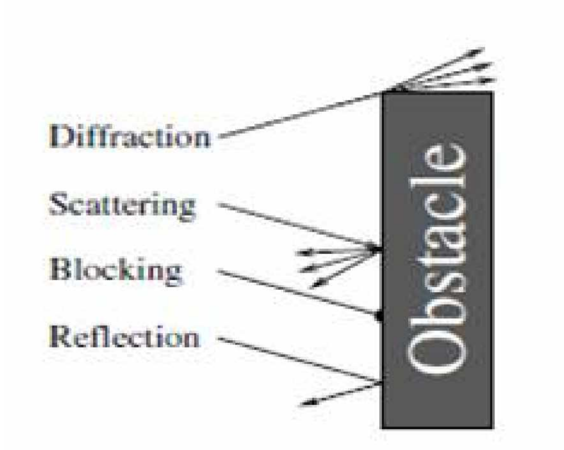


Diagram 2. How radio signals bounce off obstacles (Reichenbach et al 2006)

2.1.3 WAP, MANET, ESS Characteristics

Each wireless topology is designed to address various problems and to be implemented in different types of environments. Discussed below are the characteristics that are associated with each wireless topology and examples of the environments where they are commonly implemented.

WAP

Source to Destination path: In WAP when the source node is trying to send data to a destination node, the source first sends the data to the WAP, which then forwards it to the destination. Thus two nodes never directly communicate using this technology, as the WAP operates as the router, makes decisions on packets and forwards them on. By there only being a single WAP for all nodes to gain access to and the nature of it being a shared medium it has been identified that after 24 nodes the throughput of the WAP decreases with each additional node added (Cisco Systems Inc. 2010).

Node roaming range: The range provided by a WAP is dependent on other factors as previously discussed, such as obstacles. As reported by Cisco an optimal distance is 300 feet (Cisco Systems Inc. 2010) for a WAP, thus if users go beyond this distance they can expect to lose signal.

Implemented Environment: As a result of it having a limited distance, as noted above, WAP is thus widely deployed in small environments predominately providing 'Wi-Fi Hotspots' in coffee shops, small hotels and offices. Due to the nature of WAP an infrastructure has to exist, such as electricity source, backbone network connection and for optimal range the WAP has to be mounted on the wall (Cisco Systems Inc. 2004b).

MANET

Source to Destination path: MANET is an infrastructureless, self-configuring topology which requires no dedicated APs to route packets. It means that source to destination nodes communicate via peer to peer through either directly communicating or the routing of packets through intermediate nodes. A direct impact of being infrastructureless means that the connections between nodes may break because of intermediate nodes moving within the environment, resulting in a new route to the destination having to be established. This process adds delay to the transmission of data; in regards to VoIP and video conferencing traffic subsequently resulting in a significantly decrease in the quality of the call.

Node roaming range: Strength of MANET is its free roaming of nodes within the environment because of its infrastructureless topology resulting in the range of communication between nodes to be potentially greater than that of WAP or ESS. It is dependent on intermediate nodes being in the positions to allow a path between the source and destination node to be established. However performance significantly decreases when the hop count increases to reach the destination as illustrated in diagram 3 below. What is noticeable is the significant decrease of throughput of the blue data on the graph which represents high priority traffic, which could be VoIP or video conferencing when compared to the pink data, which is low priority.

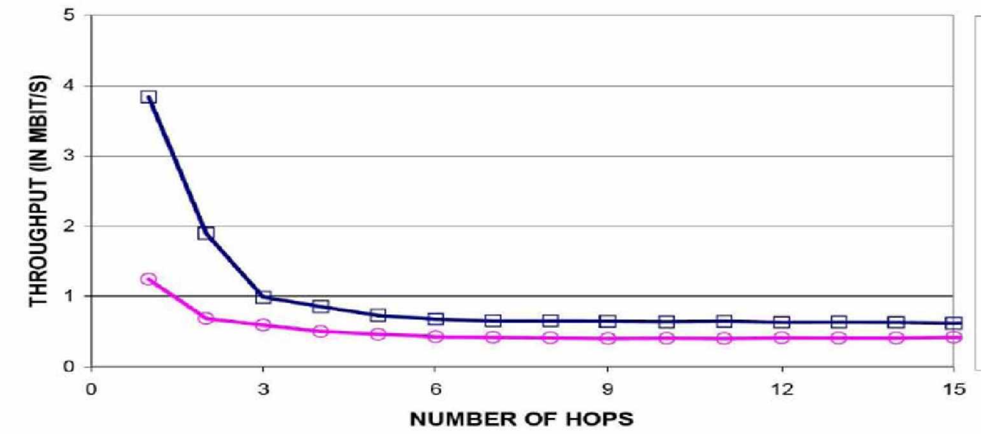


Diagram 3. Affects of Hop count on throughput for high and low priority traffic
(Brouwer et al 2007)

Implemented Environment: First designed and implemented by the military for radio communication, it is increasingly being adopted in emergency situations where no infrastructure backbone can be relied upon. (Brouwer et al 2007) The dynamic characteristic of being infrastructureless, self organising and self healing means that it can be deployed ‘anywhere at any time’ as long as the nodes are able to function in MANET mode. Although it is continued to be used in the military, it can also be used at large outdoor festivals where telecom providers’ signals are difficult to receive or after natural disasters such as earthquakes or flooding.

ESS

Source to Destination path: As described by Cisco ESS is the connection of two or more WAPs that use the same BSS’s (Basic Service Set) which are connected using a common distributed system (Cisco Systems Inc. 2004b). It extends the range of area that wireless can be provided to nodes allowing communication between different located WAP to be established. This results in two possible scenarios; the nodes are located in the same WAP area or they are located in different WAP areas. In the first scenario to establish a connection between source and destination the same process as WAP, as previously discussed, is required. When nodes are located in different WAP areas then the WAP that the source is located in has to forward the packet through the distribution system to the destination WAP area. This therefore increases the end-to-end delay of the delivery of the packet, which impacts on the quality of the call.

Node roaming range: By ESS extending the range through connected WAPs results in the nodes being able to roam in a larger area whilst receiving coverage. When a node detects a stronger signal as it becomes closer to a new WAP then a ‘handoff’ occurs and the user becomes associated with the new WAP. During this handoff period, that can last up to 300ms (Geier 2007), no data traffic can either be sent or received by the node. This short delay is an overhead and for time sensitive traffic like voice and video conferencing can be detrimental to the call quality the user experiences.

Implemented Environment: ESS extends the range of a single WAP; deployed to cover larger areas typically in enterprise networks (Cisco Systems Inc. 2004b) and university/college campuses. For example the university of North Carolina which implements 222 WAPs throughout their campus to provide complete wireless coverage (Abyzov et al 2008).

2.1.4 Conclusion

A comparable study of voice and video conferencing call capacity between each topology is not possible, as noted earlier each topology addresses a specific problem or environment thus having unique characteristics. Despite this, due to the common medium (radio waves) they all use it leads to them suffering from the same problems such as hidden node and obstacles that have a direct impact on their performance.

2.2 Analysis into the Characteristics of VoIP and Video Conferencing Traffic

VoIP and video conferencing traffic are two types of traffic that hold unique characteristics and dependences when compared to other traffic such as FTP and HTTP. The significant dependency is that both VoIP and video conferencing are sensitive to delay, which has a significant impact on call performance.

The rest of the section identifies how packet size affects performance and the characteristics of VoIP and video conferencing data flows.

2.2.1 Packet Size

The wireless medium as discussed previous is a resource that all nodes need to compete to gain access. Thus the efficiency of sending packets is critical in order to utilise this time to send across the medium specifically in regards to VoIP and video conferencing packets where delay on these packets has a critical impact on call quality. As documented in the Dublin Institute of Technology 2006 (Cranley et al 2006) experiment it was identified that small packets were less effective than larger packets due to increased lost packets from the increase of packets being sent. Due to the recommendation for VoIP and video conferencing being less than 1% packet loss (Geier 2007) small packets can be seen inappropriate for this type of traffic further justified by small packet size being identified as “inefficient channel usage” (Davis, Keegan 2006). Thus large packet size could be considered to make more efficient channel usage and time access on the medium. There is however a documented problem using this method by long packets being more vulnerable to errors when compared to short packets (Anastasi et al 2004). This as a result, specifically in regards to VoIP and video conferencing means that it is more noticeable when packets are dropped as each packet contains more data.

2.2.2 Characteristics of VoIP

The characteristics that affect VoIP traffic and directly affect the quality of the call as documented by the ITU and Cisco is delay, delay variation and packet loss (Durkin 2003). These characteristics have recommended guideline values that if not met will result in the call being dropped. Below are the three characteristics documenting their recommended values to provide a standard of call acceptable to the user and the affects if they are not met.

Delay

The delay characteristic also commonly referred to as 'end-to-end delay' (ITU 2003) is the period of time a packet takes to reach the destination node from the source node. In this case the time delay between when the caller speaks to when the called party hears it. This delay is broken down into three areas: packetization, serialization and propagation delays, which when added together result in the delay. The recommended guidelines as documented by the ITU G.114 document (ITU 2003) is 150ms or below, which has been adopted and documented as the standard by all manufacturers when referring to delay in regards to VoIP. The impact of delay above this value results in gaps in the conversation leading to two users speaking at the same time as they perceive the user as not talking. However this delay value can be extended to 200ms as VoIP calls are increasingly being made over longer distances, where 200ms delay still provides "acceptable voice quality" (Durkin 2003).

Delay Variation

Delay variation also referred to as 'jitter' (Durkin 2003), is the "difference in the end-to-end delay between sequential packets." (Hattingh 2005) The recommended value jitter should be below 150ms as beyond this value users can perceive this delay in the form of non fluent speech flow. Two contributors to this problem is varying outputs delays on routers and processing of the routers additional to the serialization delays of variable-sized packets (Durkin 2003).

Packet Loss

Packet loss in regards to Quality of Service (QoS) is packets dropped due to network congestion and not as a result of network outages or interface flapping (Hattingh 2005). Its impact on the quality of the call has a dramatic effect, with gaps in the conversation, which is directly affected by the size of packets used. The recommended value is less than 1%, which demonstrates how sensitive VoIP is to packet loss with such small tolerance (Cisco Systems Inc. 2004b).

2.2.3 Characteristic of Video Conferencing

Video conferencing uses the same characteristics of that of VoIP: delay, delay variation and packet loss, because voice and video are dependent on each other to stream synchronise during the call. Another characteristic is that video produces large bursts of data flow traffic, which can result in it consuming a large percentage of the network bandwidth (Hattingh 2005). As result codec is used to reduce this bandwidth consumption and traffic pattern for example H.323 or H.263.

2.2.4 Conclusion

The effect of the size of the packet used for time sensitive traffic as discussed is one of much research into finding the optimum packet size. Implementation of this traffic type on a medium such as wireless further increases the challenge. What has been identified is that delay, delay variation and packet loss and their respected values highlight the low tolerance that VoIP and video conferencing are able to tolerate. These identified values, which are the recommendations by ITU or the network manufacturer Cisco, will be used within the research of this report to identify the point that call quality in the network falls to a standard which is unacceptable to the user.

2.3 QoS Mechanisms

2.3.1 Background

The IP networks of the 1990s can be viewed as best-effort networks since there was no differentiation of service between applications passed across these networks. Thus in the following years when time sensitive traffic was introduced across these networks it meant that the reliability of time sensitive packets not being dropped can be described as “cross fingers and hope” (Held 2002). QoS mechanisms were first introduced in the mid-1990s when IETF first published the Integrated Services (IntServ), and since then QoS mechanisms have continued to be developed leading to enterprises being able to deploy converged networks (Hattingh 2005).

The meaning of QoS has been described in several different ways with Cisco documenting QoS as the “measure of a system’s service availability and transmission quality” (Hattingh 2005) whilst Jamalipour (2003) referring to QoS as “instead asking for a good network service, the user is asked to specifically request other sorts of measures such as connection speed or delay”. Despite the varying descriptions of QoS the end user or network administrator is likely to be the person who will identify if the expected QoS is being met (Hattingh 2005).

2.3.2 Requirements of QoS

The requirements of QoS according to Jamalipour (2003) can be classified into two main areas:

- **User-Level requirements** Since the end user is likely to be the person who identifies if the QoS does not meet their requirements user-level requirements are broken down into three areas called: criticality, cost and security. As the user perception of QoS is based on previously observed behaviour (Hattingh 2005), it means not all users use the same factors when evaluating QoS. For example, some users might find it acceptable to view poor video quality as long as they have a high quality of sound, whilst others may disagree. It is therefore important that clear definitions of what users define as acceptable and unacceptable exist. Cost is another important factor for the user, as it defines how the user will pay the Internet Service Provider (ISP) in order to receive the QoS they require. For example the ISP may charge per use of the service (VoIP or video conferencing traffic) or per unit, which in regards to VoIP could be charged per minute of usage. Security in regards to QoS aims to ensure that the QoS data is treated at a level of service that is required and thus as a result of poor security the user may experience performance below that expected perhaps due to malicious attacks.

- **Technology and Network requirements:** To provide the user-level QoS requirements the technology and network requirements have to be in place by the ISP, which Jamalipour (2003) classifies into three areas: bandwidth, timelines and reliability. The bandwidth is the speed or the data rate available to the user application; it is commonly viewed as a simple solution to improve QoS by increasing the bandwidth. This however only provides the bandwidth speed between the customer’s premises and the local exchange and not the internet backbone. Timelines which incorporates delay, response time and delay variation is a critical factor especially in regards to time sensitive applications which have a delay threshold before packets are discarded due to being out of sync with the rest of the application. The reliability directly links to Cisco definition of QoS; service availability and transmission quality. Ensuring that the network does not fail in regards to downtime but also the delivery of packets is a critical factor in ensuring QoS, with Cisco recommending achieving a system uptime of 99.999% (Hattingh 2005).

2.3.3 QoS Models

With converged networks allowing multiple network services to be sent over a single network infrastructure as discussed above leads to QoS mechanisms being developed. Two QoS models are Integrated Services (IntServ) and Differentiated Service (DiffServ), which differentiates levels of service between network services that the original 'Best Effort' model (Wallace 2008) could not provide and are discussed below.

- **Best Effort:** The best effort model does not provide any QoS and is the original model for the internet and to this day continues to be the default configured on networks. Despite not supporting QoS and therefore relying on protocols to guarantee packet reliability as previous mentioned "cross fingers and hope" (Held 2002) it is a highly scalable model requiring no specialist equipment or software.

- **IntServ:** Provides QoS to services, guaranteeing delivery and predictable behaviour over the network through implementing Resource Reservation Protocol (RSVP). Before the service is available to the user (for example VoIP call), during the connection establishment between the source and destination node's RSVP tries to reserve resources assigned to the service requirements on each device along the connection path. Through reserving resources along the connection and intelligent queuing mechanisms on each device, this is how guaranteed delivery of packets and expected behaviour can be achieved. So when high traffic bursts occur it does not affect the performance of the service that is established. If the reservation of the resources on any of the devices along the connection is not possible it results in the user service is not started. Through the explicit reservation of resources along the connection path, as identified by Cisco (Wallace 2008) and Abbas and Villalba (2003) makes it a non scalable solution when considered its implementation over the size of the internet. In addition to the resource reservation the continuous signalling to ensure the QoS of the link, adds additional network traffic and when scaled to the size of the network, the internet, it further justifies how it is not a scalable solution.

- **DiffServ:** Developed to overcome 'no QoS' supported by best effort and the lack of scalability of IntServ. It classifies service packets into different classes; differentiating the service QoS requirements by setting the ToS (Type of Service) field in the IP packet header. As the marked packets enter the DiffServ network Domain (group of routers implementing DiffServ policies) the packets ToS field is analysed or over written if tighter control in regards to QoS is required. The routers then queue and routes the packets based on the routers implemented queuing and scheduling algorithm (discussed below). What makes it a scalable solution is the QoS is managed on per hop basis allowing interoperability of vendor devices running an IP network and the non reservation of resources (Hattingh 2005). The disadvantages of implementing DiffServ is the lack of per service flow control and guaranteed delivery level of service that IntServ can provide.

2.3.4 Queuing and Scheduling Algorithms

The role of scheduling algorithms refers to "set of features that determines how a frame, cell, or packet exits a device." (Hattingh 2005) The common device used in IP networks that implement queuing and scheduling algorithms are routers. They contain memory buffers that allow for temporary storing of frames, cells or packets before being scheduled when there is congestion on the output interface. These buffers contain queues that order the frames, cells or packets linked to a specific interface, which depending on the scheduling algorithms may contain several virtual queues associated with a single interface. There are several scheduling algorithms that provide different performance to traffic on the outgoing interface of a link depending on the QoS required for the traffic. Below is a list of commonly implemented scheduling algorithms.

- **First In-First Out (FIFO):** Implemented in IP networks that do not implement QoS and commonly the default setting on routers (Opalsoft 2009), FIFO is a scheduling algorithm that only has one queue thus treating all traffic classes equally. Its scheduling algorithm works, as the name would suggest, on the principle that the first packet that comes into the output interface queue is the first packet that goes out the interface. Once the buffer queue is full then new packages entering the back of the queue are dropped.

- **Priority Queuing (PQ):** Consisting of only four queues, classed as high, medium, normal/default and low, PQ is regarded as being able to provide real-time traffic with a high level of QoS (Wallace 2008). Each of the four queues implements FIFO queuing, with the PQ scheduler not moving onto the lower queue until the current queue is empty. This means that PQ can starve lower priority class queues if there is a continual flow of high priority traffic.

- **Weighted Fair Queuing (WFQ):** WFQ classes traffic into different queues depending on the ToS field in IP header. Each queue is equally weighted through dividing the bandwidth of the output interface by the number of flows (Hattingh 2005). To ensure that all queues are allowed time to the medium WFQ implements a round robin scheduling method that ensures that the different flow queues are served in a circular manner, thus preventing any queues from starvation. The QoS bandwidth guarantee per flow is not possible, which CQ offers (discussed below), instead WFQ bandwidth allocation constantly changes as flows are added or ended (Hattingh 2005).

- **Custom Queuing (CQ):** CQ classifies packets into 17 classes according to the rules set regarding to the CQ policy. Queue 0 is a system queue (for example keepalives and routing updates), which cannot be changed leaving 16 distinct queues available for user queues. For the number of active queues each is assigned a byte count of the interface bandwidth, thus ensuring mission-critical applications get sufficient bandwidth but also ensure non mission-critical applications obtain a percentage of the bandwidth as well (h3c 2010). Similar to how WFQ operates CQ implements a round robin scheduling method to circulate evenly between the different queues. The weakness of CQ in regards to real time traffic is that is it not capable of guaranteeing strict priority, which PQ can.

Performance Comparison of Queuing and Scheduling Algorithms

There have been several studies for example Asif and El-Alfy (2009) and Velmurugan et al (2009) based on the performance analysis of scheduling algorithms. The results of these studies aimed to demonstrate under different network traffic loads how the performance of each algorithm performs.

The results from Asif and El-Alfy (2009) report which was a comparable study of scheduling algorithms consisting of FTP, voice and video traffic flows is shown in diagram **4,** below. What is noticeable is FIFO has a significantly higher delay compared to the other algorithms, this can be noted as a result of the other routing algorithms priority scheduling of time sensitive traffic such as voice and video conferencing reducing the combined average queuing delay.

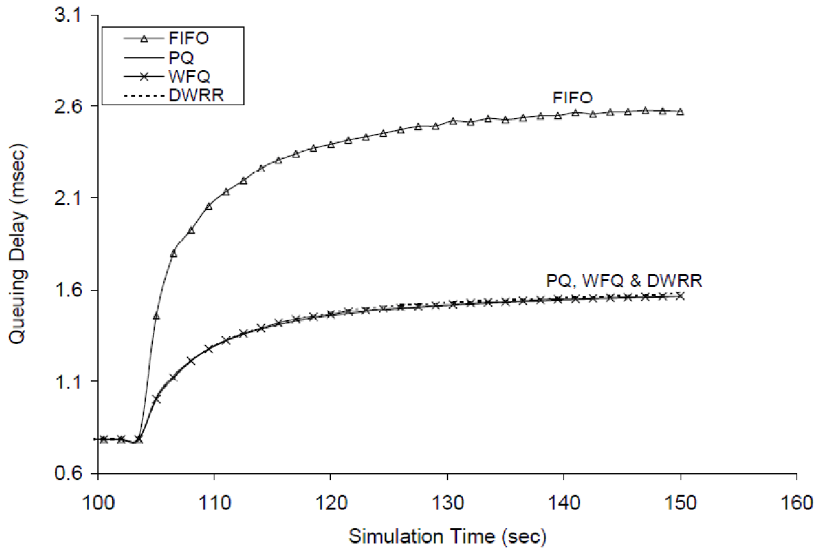


Diagram 4. Queuing delay performance of different queuing algorithms (Asif, El-Alfy 2009)

Velmurugan et al (2009) comparison study of queuing disciplines shown below in diagram 5, used the same algorithms as in Asif and El-Alfy (2009) but implemented additional queues by modifying PQ, WFQ and DSCP. PQ and WFQ were modified and called ProtocolbasedPQ and ProtocolbasedWFQ so that it implemented random-early drop (RED) thus when congestion is detected random packets are dropped to prevent further congestion. Since it is not possible to implement RED on DWRR, Differentiated Services Code Point (DSCP) is implemented to WFQ, which its purpose is the same as ToS field in an IP header. However as documented on a website called Networld (2010) DSCP is phasing out the traditional ToS header. DSCP uses the same field however further utilises the ToS field by using the first 6 bits of the 8 bit field proving 64 different values compared to the traditional ToS of 3 bits and 8 different values (Rhyshaden 2010). Thus through the implementation of DSCP, means that greater differentiation between QoS service classes can be achieved and further provides QoS as time sensitive traffic puts greater demands on our networks. WFQ and ProctocolbasedWFQ noticeably has the highest end to end delay due to its fairness scheduling policy of round robin and as a result voice is not provided with enough access time to the medium due to other traffic streams such as FTP and video. As can be expected PQ and ProtocolbasedPQ performed the best due to it not moving onto the next queue until higher priority queues are completely emptied.

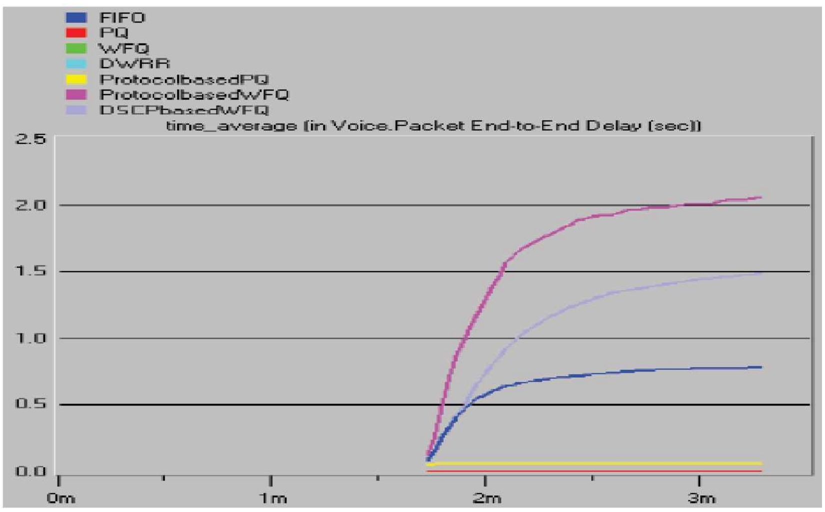


Diagram 5. Traffic Received for Video Conferencing for different queuing algorithms (Velmurugan et al 2009)

2.3.5 QoS challenges over Wireless

The convergence of network services have driven both wired and wireless transmission medium networks to implement QoS mechanisms to provide appropriate service quality to each class of service. The internet is constructed of wired networks with the development and the nature of cable as the medium means that its QoS challenges vary from that of the wireless medium. There are several challenges in developing QoS mechanisms for the wireless medium as discussed in Moterola (2006) whitepaper and Marwaha et al (2001) report, and some of these are discussed below along with how they are not explicable to the wired networks.

- **Capacity Constraints**: The wireless medium bandwidth is a “scarce and expensive” (Marwaha et al 2008) resource, which if 802.11g is implemented provides up to 54Mbs bandwidth, approximately half that of a commonly deployed wired 100Mbs Ethernet connection. Since WAPs share the medium, it means the bandwidth has to be divided equally between all the nodes associated with the WAP. As new nodes arrive and use the WAP the bandwidth allocation to nodes is further reduced, ultimately leading to the saturation of the WAP. The challenge wireless QoS mechanisms have to address is to ensure services are provided with the correct QoS under potentially low bandwidth WAPs.

- **Unreliable Communication Medium:** As discussed in section 2.1.2 the nature of wireless medium of Radio Waves (RW) makes it exposed to a large number of outside factors that can affect the reliability and performance of the wireless communication; including weather, interference and wall thickness. Additional factors are multi-path fading (Marwaha et al 2008) and as documented in Petcher (2006) wireless suffers significantly more data being lost during transmission due to the data becoming corrupted when compared to wired transmission. These factors are dynamic and change with the environment that the WAP is deployed in, thus creating a QoS mechanism that can adapt to the changing characteristics of the environment is extremely difficult. In order to more accurately estimate and manage these factors, software such as Mentum (2010) and Airmagnet (2010) have been developed to accurately analyse wireless performance in the environment before deployment. It is however not a QoS mechanism rather a tool that can be used to identify what equipment may be required to provide wireless coverage and QoS in the environment.

- **Node Mobility and Dynamic Topology:** As discussed in this report the introduction of node mobility into the wireless environment brings several new challenges, which as result impact on being able to provide QoS that is applicable to wired networks. Due to the dynamic nature of nodes in environments such as MANET, means that routing information can become out of date relatively quickly and further increased as node mobility speed increases further discussed in section 2.5. This results in creating a connection between source and destination nodes becoming more challenging, before the QoS mechanism is implemented, which may require being re-established if the connection drops. Providing this type of QoS has limitations, as once the mobility reaches a threshold in which the topology changes happen quicker than the QoS can be re-establish it will not only no longer be functional but neither will a route to the destination be established.

2.3.6 QoS Solutions over Wireless

To address the additional challenges that wireless networks have when developing and implementing QoS mechanisms several solutions have been developed, and these are discussed below.

Quality Of service RoUting in Wireless Mesh (QUORUM)

This QoS solution is a reactive routing protocol that aims to provide a "QoS constrained route from source to destination." (Kone et al 2007). To achieve this it uses three parameters (1. minimum bandwidth, 2. end to end latency and 3. link robustness), which are evaluated along the path of the intermediate nodes between source to destination to ensure the QoS requirements are able to be met but not reserved and thus a decision can be make as to whether the connection should be established. To gather the parameter values which decisions are based on, nodes or WAPs send periodic control data messages. The robustness of a link is calculated through the number of successful 'HELLO' packets received from a neighbour during a specific time window, which is then added to the number of previous successful 'HELLO' time window periods. The effectiveness of this QoS routing protocol is shown in diagram 6, below, which compares the performance of 5 flows when implementing QUORUM and AODV (AODV provides no QoS mechanism) in the same environment. As can be seen the flow between F:27-28 in regards to QUORUM implementation does not have delay because the flow is rejected as under the QUORUM routing policy the flow requirements could not be achieved along the route path.

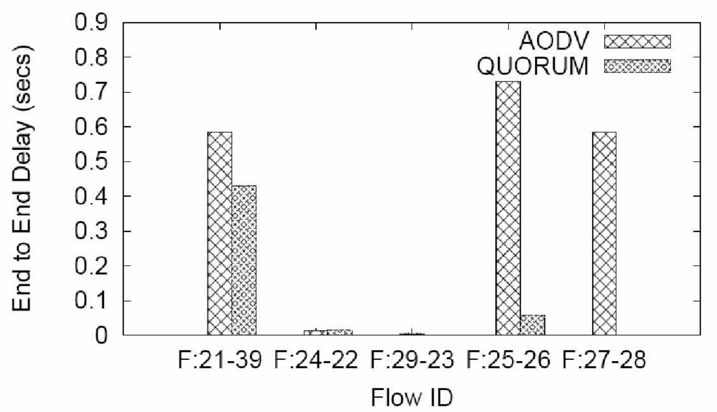


Diagram 6. End to End delay comparison between AODV and QUORUM for five flows (Kone et al 2007)

Since being a reactive protocol it makes QUORUM a scalable solution, unlike IntServ which implements its QoS during route establishment however in the case of QUORUM it does not reserve bandwidth for the application flows along the route devices. Its performance as the number of flows increases compared to AODV in diagram **7** below, shows that on average the delay is 33% lower.

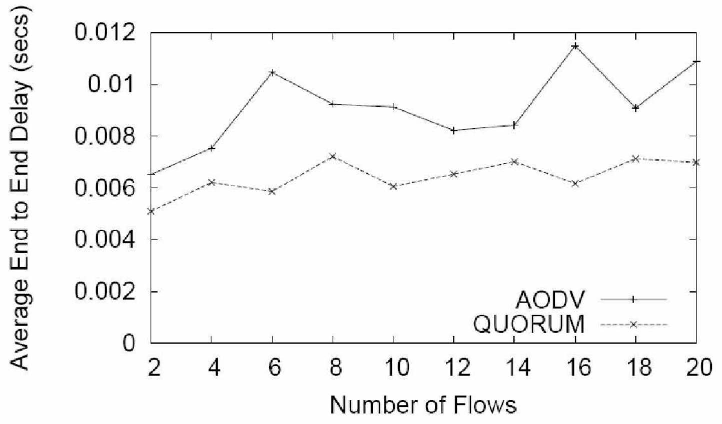


Diagram 7. Average End to End delay as number of flows increases (Kone et al 2007)

QUORUM despite aiming to select the most robust route the routing protocol still suffers from high node mobility, this causing increased topology changes in the environment. Due to it being a reactive protocol means that when a route break occurs QUORUM has to re-establish the route if there is one available that meets the flow QoS requirements. As previously discussed, ultimately when node mobility is at a state that topology changes are occurring more frequently as the routing protocol can establish routes to destination nodes, QUORUM implementation will experience decreased performance as a result.

QoS Enhanced Basis Service Set (QBSS)

The development of the IEEE 802.11e standard has introduced a new element called QBSS, which is a beacon frame containing information on current traffic state relating to the QoS enhanced Access Points (QAP) (Coskun et al 2008).

The beacon frame contains three parameters (Coskun et al 2008):

- **Station count**: Number of nodes currently associated with the AP.
- **Channel utilization**: The percentage of time that the channel is determined to be busy.
- **Available admission capacity**: The amount of time that can be used by explicit control; identifying if there are enough resources available to accept the connection for an application.

The information that the beacon contains in regards to these three parameters can be used by both the AP and nodes. In regards to AP, the beacon information is used to identify if the AP can accept anymore admission control requests to reserve resources for applications. The node not only uses this information to decide which AP to associate with when multiple APs are available, but the beacon also provides information to the node as to weather the AP can meet the demands that the node application requires. This QoS mechanism is scalable since it is not dependent on route paths between source and destination nodes, but only provides a limited level of QoS identifying to nodes and neighbouring APs if their QoS demands can be met.

2.4 Analysis of QoS in Regards Supporting VoIP and Video Conferencing Traffic

As previously discussed the increase in demands being put on wireless networks has resulted in time sensitive traffic, such as VoIP and video conferencing being introduced onto this medium. This convergence of data across the medium has lead to QoS being required to distinguish between different types of traffic in order to provide the level of service each application requires. To provide this QoS, an enhancement to the MAC layer of the 802.11 standard was produced to develop the 802.11e standard. By enhancing the MAC layer means that it is not dependent on the physical layer and as a result it can be implemented on different physical layers technologies. QoS is defined as "the measure of a system's service availability and transmission quality" (Hattingh 2005) Due to the characteristics of VoIP and video conferencing as discussed in section 2.2 means that QoS is critical to provide the level of service required for these applications.

2.4.1 802.11e EDCA

One 802.11e method that can be supported across all three wireless topologies as documented in this report is EDCA (Enhanced Distributed Channel Access), which is an enhancement of the DCF MAC protocol supporting multimedia applications with QoS requirements. (Berg et al 2006) A key characteristic in EDCA compared to 802.11b is EDCA allows a maximum of four queues for the different traffic types to be put into. Distinguishing data types is achieved through Access Categories (AC) or by source and destination IP addresses, which as documented in Andreadis et al (2006) conference paper using AC method, the priority of traffic is VoIP, video, best effort then background, with VoIP being the highest priority and the lowest being background (Andreadis et al 2006). The mechanism that ensures higher priority access to the media over other queues is through the parameter value of the 'minimum contention window' being lower than other queues. Other parameters that are required to be configured are Arbitration Inter frame Space (AIFS), Transmission opportunity limit ($TXOIP_{limit}$) and maximum Control Window (CWmax). The values assigned to these parameters dictates the performance of QoS the node will experience and research identifies that incorrect values can "significantly degrade the QoS performance" (Andreadis et al 2006)

2.4.2 Impact 802.11e on Performance

Improvement to performance

As would be expected from implementing QoS mechanisms it does increase the QoS service to higher priority traffic such as voice and video, as demonstrated in diagram 8 below taken from the 2006 report by Andreadis et al (2006), which was a comparable study of varies types of traffic using 802.11b and 802.11e. Diagram 8 clearly demonstrates that 802.11e provides a constant level of delay and significantly lower delay with no irregular fluctuations when implemented using 802.11b. Thus due to the constant level of delay means that the quality of the call the user will experience when implemented using 802.11e would be one that would adhere to the ITU G.114 recommendation.

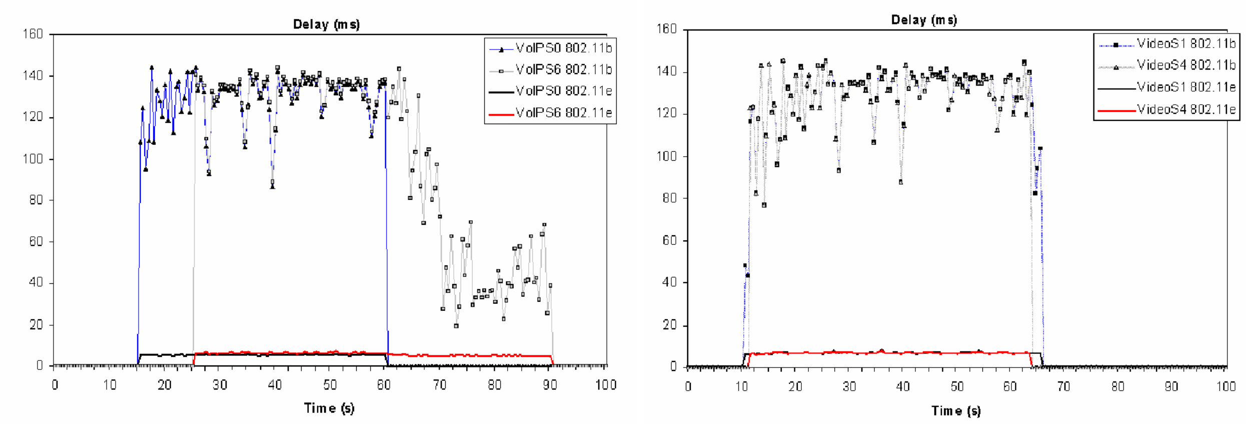


Diagram 8. The delay of voice and video when implemented using 802.11b and 802.11e (Andreadis et al 2006)

Challenges of implementation

Despite the improved performance that implementing 802.11e QoS mechanism can have on traffic, there are also challenges and problems as discussed below.

As identified in the 2005 journal "Quality of service provisioning in 802.11e networks: challenges, approaches, and future directions" a trade-off has to occur between QoS techniques and energy consumption (Dey et al 2005). Thus due to mobile nodes having a limited battery life means that conserving energy is a critical area for these nodes. Implementing 802.11e can as a result reduce their battery life time due to potential longer waiting times to send the data due to four priority queues being introduced and nodes not being able to go to sleep due to scheduling schemes. They propose that a technique is required to be developed that accounts for the device limitations in regards to battery by incorporating sleeping scheduling on queues (Dey et al 2005).

When 802.11e is implemented on different network topologies it has been documented that it causes different effects on performance. Whilst being deployed on a WAP and ESS, the problem identified is that the downstream direction of the WAP becomes bottleneck.(Berg et al. 2006) It is the result of the WAP being the device that all traffic has to be routed through, causing potential starvation of lower priority traffic. To prevent this occurring it is suggested by the report that EDCA parameters are changed from that of the nodes because the demands on the WAP being different from the client nodes (Berg et al 2006). Ad Hoc infrastructureless topology means that providing end-to-end QoS is affected by many variables such as the number of active neighbours a node has and number of hops from source to destination (Brouwer et al 2007). Its dynamic nature means that there are variables that 802.11e can take into account and as a result it is Brouwer et al (2007) view that "ad-hoc networks resulting in end-to-end QoS is only sufficient for a limited number of mildly loaded hops".

2.4.3 Conclusion

As documented increasingly networks are becoming converged to provide a complete enterprise solution over the one network medium. To address this convergence the deployment of 802.11e as discussed has a significant positive impact in delivery of delay sensitive traffic such as voice and video conferencing. Despite this positive impact the nature of the wireless topology does have an impact on the level of QoS that can be provided and is out with the control that 802.11e aims to addresses.

2.5 MANET Protocol

Traditional routing algorithms are described as "aimed at finding optimal routes to every host in the network" (Boukerche 2001). Despite this, distance vector and link-state routing algorithms, are not suitable for Ad Hoc and MANET networks as noted by Boukerche (2001) and Objectives (2006), and thus require specially designed Ad Hoc routing algorithms. Boukerche (2001) recognised that the host devices within AD Hoc and MANET environments are not dedicated routing devices, which subsequently limits the device resources availability to perform routing.

2.5.1 Aim of Ad Hoc Routing protocol

To address the limitations of Ad Hoc devices and the environments they are deployed in, Objectives (2006) states the characteristics that an Ad Hoc routing protocol must incorporate:

- **Discover source to destination path**: There has to be a mechanism to discover a route from the source node to the destination node through the use of intermediate nodes.

- **Maintain the route path**: To retain connection between source and destination nodes when failure or changes in node position occur that result in a link break.
- **Define a mechanism to exchange routing information**: As in traditional routing, exchanging routing information is required to obtain routing tables or set routes to reach the destination node.
- **Nature of wireless**: The nature of wireless as discussed earlier in section 2.1 has to be considered, such as node mobility, signal strength and interference as they affect the conditions that the routing protocol has to operate under

2.5.2 Challenges faced in MANET routing protocol

As documented in AlTurki and Mehmood (2008) there are several challenges faced when routing protocols are implemented into MANET environments due to some of the problems discussed below:

As documented in AlTurki and Mehmood (2008) there are several challenges faced when routing protocols are implemented into MANET environments due to some of the problems discussed below:

- **Mobility/topology changes**: The nature of MANET and the movement of nodes mean that nodes can create new and end neighbour relationships with other nodes, thus creating new route links but also the breakage of links. This is a well documented problem that makes providing QoS routing difficult as documented in Rubin and Liu (2003) further supported by Agrawl et al (2008) suggesting from their study that "traffic pattern, node density and initial pattern of nodes effect the routing performance". The speed of nodes and the impact on routing performance was studied by Klein (2008) and it was identified that as node speed increased link breaks increased as a direct result. This is shown by the end-to-end reliability in diagram 9, below, where performance is reduced when node speed is increased when implementing AODV, SBR and OLSR Ad Hoc routing protocols.

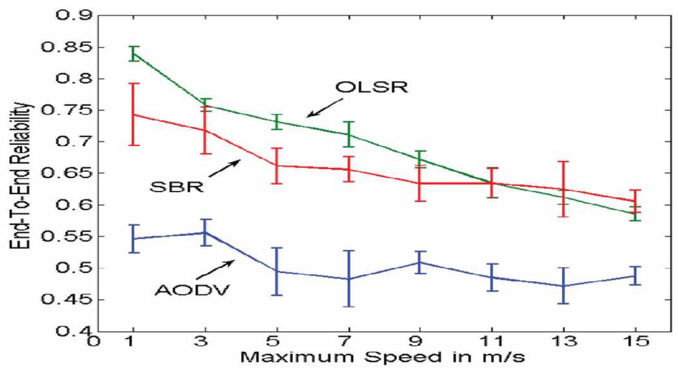


Diagram 9. End-to-End Reliability in relation to increase in node speed (Klien 2008)

A method developed and tested by Rubin and Liu (2003) tried to improve end-to-end reliability by routes only being selected when the lifetime of the link had passed a threshold period of time, thus the link can be seen as stable. They concluded when their mechanism was implemented to On-Demand Distance Vector (AODV) that it improved its performance by reducing the number of route request packets being sent over unstable links.

- **Energy efficient algorithms**: As mobile devices have limited energy life it is critical that the routing protocol does not make inefficient use of the energy. A level of energy consumption and routing performance has to be met to ensure that each demand is met.

- **Security**: At present security is not widely deployed and has thus “lacked much attention” (AlTurki, Mehmood 2008). With MANET having no infrastructure backbone securing the whole environment is challenging not just in regards to routing protocols, but is an area that in the future that will have to be further incorporated into current or new MANET routing protocols.

2.5.3 Categories of Ad Hoc routing protocols

MANET routing protocols as documented in Shrestha and Tekiner (2009) can be classified into three categories, as illustrated in diagram 10 below.

The rest of this section will discuss reactive, proactive and hybrid protocols, with a comparison between reactive and proactive performance.

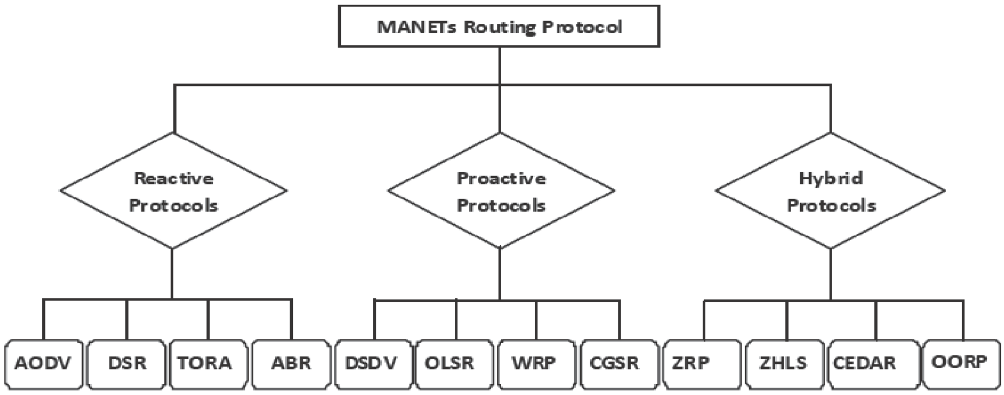


Diagram 10. MANET Routing Categories and Protocols (Shrestha, Tekiner 2009)

Reactive Protocols

Reactive protocols are also commonly known as ‘On-Demand’ routing can be described as ‘pure on-demand route acquisition” (Perkins, Royer 1999) because nodes “do not lie on active paths neither maintain any routing information nor participate in any periodic routing table exchanges” (Perkins, Royer 1999). Thus when a node is required to communicate with a destination node a route is created through the route discovery process and once established it is maintained until the destination becomes inaccessible along the specific path from the source or is no longer required. As diagram 10 illustrates there are several reactive routing protocols such as DSR, TORA and ABR but the widely deployed protocol is AODV.

AODV mechanism establishes a route through the process of broadcasting RREQ (Route Requests) packets through the nodes in the network until a node has a route to the destination or the destination is reached. At this point a RREP (Route Reply) which is a unicast packet is sent back along the route, which is identified from the RREQ packet information to create a link connection. Through the use of sequence numbers AODV is able to provide a loop free network by old routes to destinations not being used if the arriving RREQ packets has a higher sequence number than that in its routing table. As discussed links breaks can occur due to node mobility, thus when a link break is identified a RREP packet with an infinite metric will be sent down the route. Thus the source will either have to re-establish the route through broadcasting RREQ message to neighbours or if ‘Local Route Repair’ mechanism is enabled allowing a node along the link to try and repair the link to the destination node itself. Through being on-demand routing means that there is initial delay from having to establish a route to the destination compared to proactive protocols (discussed below), in which routes to destinations are already known but can reduce the amount of routing packets being propagated through the network.

Proactive Protocols

Another type of routing protocol is proactive and is a table driven mechanism, which attempts to maintain up-to-date routing information about all nodes in each node's routing table (Chai-Keong 1999). Its approach is that similar to traditional routing of Ethernet networks through the convergence of routing tables and period updates, reflecting the changes in the network. However as identified in Boukerche (2001) paper, these period updates and potential routing tables containing hundreds or thousands of nodes can be an expense in relation to node resources, through only a subset of node routes likely to be used. Examples of proactive routing protocols are OLSR, WRP, CGSR but a commonly used one in performance studies such as Kulkarni and Rao (2008) and Boukerche (2001) is Destination-Sequenced Distance-Vector (DSDV).

DSDV is a proactive routing protocol based on Bellman-Ford's routing mechanism, which is also used in Routing Information Protocol (RIP), but improvements have be made to remove the possibility of routing loops occurring (Royer, Chai-Keong 1999). Each node in the network maintains a routing table containing routes to all nodes using a sequence number assigned to each route to identify old route advertisements, thus preventing routing loops. The advertisement of route updates to neighbours help reduce the prevention of large overhead updates, DSDV implements two update methods: full dump and incremental. The full dump update occurs when the whole table is advertised to its neighbours through the network potentially consuming large bandwidth and propagation time, whilst an incremental update only advertises changes since the full dump, thus aiding in reducing bandwidth consumption and propagation time. Despite having two update methods, a study by Kulkarni and Rao (2008) about DSDV performance in regards to node mobility identified that due to its nature of periodic routing table updates the delay produced makes it unstable in regards to movement. They further recognize the limitations of DSDV in relation to the scaling overhead growth with the formula of '$O(n^2)$'.

Hybrid Protocol

The hybrid protocol combines proactive and reactive routing as discussed above. A node contains a limited number of proactively learned routes; however when a route needs to be discovered reactive flooding using RREQ is used to establish a route between source and destination.

2.5.4 Comparison between Proactive and Reactive protocols

There have been several studies carried out in the comparison between proactive and reactive protocols identifying that each protocol has strengths and weaknesses depending on the environment it is implemented in.

Kulkarni and Rao (2008) study concluded that AODV outperformed DSDV in regards to packet delivery ratio when 'random walk' mobility was implemented. This was identified because DSDV routing table only contains one route per destination and as a result packets are dropped if MAC layer is unable to deliver the packet. As can be seen in diagram 11, below, as the pause time increases DSDV performance increases, as a result of less mobility causing less link breaks and propagation of routing packets. ADOV can be seen to have the highest delivery ratio due to as link breaks occur a new route can be established to the destination quicker through broadcasting RREQ packets or 'Local Route Repair' rather than DSDV routing table updates.

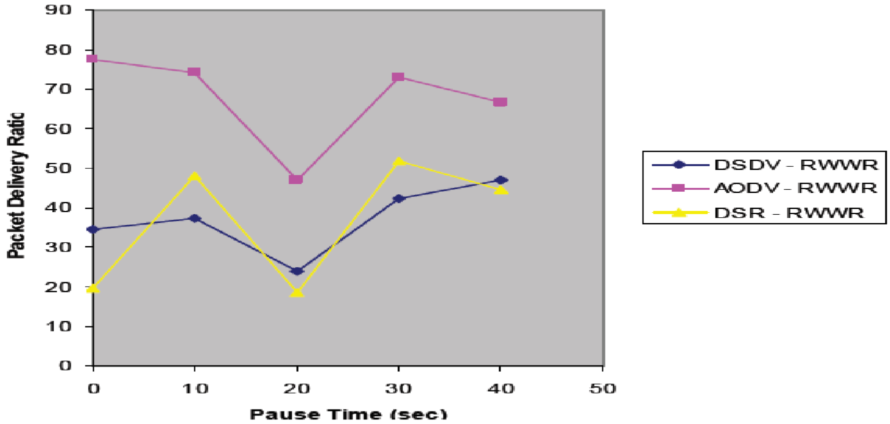


Diagram 11. Packet delivery ratio on Random walk with reflection mobility (Kulkarni, Rao 2008)

Another study by Boukhalkhal et al (2008) using 'random waypoint' mobility pattern, 10 connections with 4 packets/sec and a packet size of 512 bytes tested how node mobility count affects the end-to-end delay when implementing MANET protocols AODV, DSDV and CDRP. As shown in diagram 12, below, CBRP (Cluster-based Routing Protocol) performed the lowest but it can be seen that all performances decrease in relation to delay as nodes increase. This potentially resulting due to the environment becoming denser with nodes thus the route between source and destination involves more intermediate routes subsequently causing increase in the delay. It should be noted that this in not discussed within the literature review.

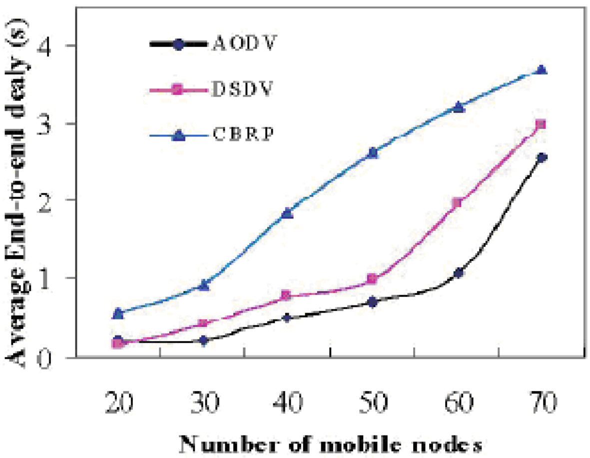


Diagram 12. Average End-to-End delay in relation to number of mobile nodes (Boukhalkhal et al 2008)

The varying levels in performance demonstrated above illustrates the point made by Boukerche (2001) that there is no routing protocol that is superior to another. Therefore when choosing a MANET routing protocol their characteristics and previous reported performance has to be considered in relation to the environment it will be implemented within.

2.6 Identify characteristics of node mobility and density

The nature of the wireless medium means that it provides the ability for the user to roam within the environment. As a result mobility and the density of nodes within the environment have an impact on the performance experienced by the nodes. Thus the following section identifies the characteristics of node mobility and density in regards to the impact of incorrectly modelling node movement for network simulation testing.

2.6.1 Node Mobility

The ability to recreate user movement in network simulation environments is one which has been the subject of much documented research. The inaccuracy of the user movement has two impacts specifically in regards to MANET, link breaks and increase traffic, which then has a direct impact on the test data produced (Belding-Royer et al 2004). The first problem, which is link breaks, is that packets are lost during this time and increases delay due to a new route having to be established. As the size of the path between source and destination nodes become larger the impact of the link breaks increases. The second impact is increased traffic because of control packets being sent more frequently due to nodes establishing and ending connections with neighbours as they move through the network. Despite mobility having a potential negative impact on time sensitive traffic, Grossglauser and Tse (2002) concluded that mobility for delay-tolerant application can in fact increase throughput. Node mobility in WAP has less of an impact due to it having a dedicated device to route traffic through. As the node moves further away from the access point (AP) the strength of the signal decreases ultimately leading to no signal being received. In ESS, a node experiences the same distance effect problem as WAP but when the node becomes closer to another access point then the connection is handed over to the new AP.

Two characteristics of node mobility are node speed and their pattern of movement, which are described below and identify how previous simulation experiments vary significantly from how users move in the ‘real world’.

Speed

The speed the node moves through the environment should reflect that of a user in the ‘real world’, however a 2005 study found that the default parameter values used in mobility modelling are often too aggressive. They found that simulation software package ‘ns2’ assign node of speeds 0-20 metre per second, which compared to their ‘real world’ experiment found users walked at an average of one metre per second. Varying speeds of up 20 metres a second when modelling user movement directly impacts on the test data and as discussed will cause increase link breaks, causing more control data and delay. To reduce the impact of fast node movement within an environment a 2001 study by Royer et al concluded that as mobility speed increases longer transmission range should be implemented by nodes. They found that from implementing longer transmission range nodes compared to short ones means that nodes were able to keep their neighbouring nodes for longer thus reducing link breaks (Melliar-Smith 2001).

Movement pattern

The movement of users within an environment when using simulation software is often implemented using random walk and random waypoint. However there is significant research that concludes that they all have flaws making them less realistic, ultimately not addressing the point that users do not walk randomly, they walk a specific path to reach a destination (Almeroth et al. 2003). The problem identified by implementing random walk, which has been called “memoryless mobility pattern” (Boleng et al 2002) due to its lack of knowledge of past locations and speeds. This results in nodes stopping and making sharp turns suddenly due to not knowing its previous state, which is referred in the 1999 study by Chiang et al. and supported in Boleng et al 2002 study. The random waypoint mobility pattern is documented to not represent user spacing and distribution in the environment as at the start of the simulation the nodes are randomly placed within the environment (Camp et al 2002). The effect of this and thus affecting the test data is that nodes located with a large number of neighbours at the start of the simulation may experience a higher performance than expected.

A 2005 study by McNett and Vowlker of wireless usage within a real university identified that of 275 users on a university campus on average only 11 % of users are in motion at any one time. The impact this has means that a large percentage of users can remain static within a campus environment at a specific point in time thus meaning link breaks or handovers, for example, are likely to occur less frequently.

2.6.2 Node Density

The movement of nodes within the environment can lead to a large numbers of nodes being located close to each other within the environment thus having an impact on node performance in all wireless topologies. In regards to WAP and ESS, Cisco recommend that no more than 15-20 nodes should be associated with a single AP (Cisco Systems Inc. 2004b). This is a result of the characteristic of the wireless medium only allowing a single node or AP accessing at anyone time. Beyond 20 nodes it would be expected that nodes would not get access to the medium due to the high number of competing nodes. MANET topology however relies significantly on node density in order to function due to its lack of centralized distribution and infrastructure, thus resulted in an optimum number of neighbouring nodes a node should being identified. A 1978 conference paper by Kleinrock and Silvester identified that a node having six neighbours was required for optimum throughput on a static node network. A more recent study in 2001 (Melliar-Smith 2001) challenged this and identified that seven to eight neighbours was the optimum number, with the introduction of node movement requiring higher connectivity. Diagram 13, below, identifies the probability of a node finding a route to the destination in relation to the number of neighbours associated with the node.

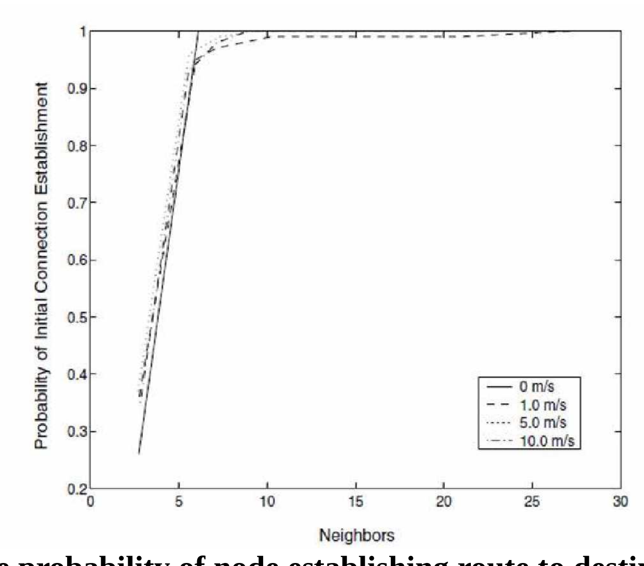


Diagram 13. The probability of node establishing route to destination in relation to neighbour count (Melliar-Smith 2001)

With an environment there can be a point at which the population of nodes to the physical area causes a decrease in performance as a result of network saturation. The point in which the performance decrease is unique to each environment, thus as a result of the characteristics of each environment being unique, such as physical size and obstacles. This is why a formula has been created and documented in a paper by Grossglasuer and Tse (2002), which specifies that throughput per source-to-destination pair decreases approximately at $1/\sqrt{n}$, where ‘n’ is the number of nodes per unit area. The decrease in performance is the direct result of the increase in the number of collisions occurring within the network, resulting in retransmission and in relation to VoIP and video conferencing means that call quality significantly drops (Melliar-Smith 2001).

2.6.3 Conclusion

It can be seen from what has been discussed that node density is the direct result of node mobility patterns. This is why the accuracy and realism of node movement specifically in regards to network simulation is critical to produce test data that is accurate and a fair reflection of the environment and node characteristics. This is demonstrated from McNett et al (2005) study of real student movement on their university campus identifying user movement and patterns are both slower and less frequent as default parameters are used in network simulation software. The infrastructureless topology of MANET means node density places a critical role in the level of performance that can be achieved from this wireless topology, providing both a positive and negative impact.

2.7 Analysis of Call Capacity of each Topology

Indentifying the call capacity of VoIP and video conferencing means that an enterprise is able to identify if a wireless solution is able to provide the call capacity that they require. From research there has been significant studies into this area in regards to VoIP over wireless, however in regards to video conferencing it appears that only two papers have been published in this area. Thus as a result of this report it will be the first to identify the call capacity of video conferencing over the three different wireless topologies. As discussed in section 2.1.3 each topology has unique characteristics and thus each call capacity will vary as result.

Discussed below is how call capacity is identified and from previous documented research the call capacity in the three different wireless topologies.

2.7.1 Identifying Call Capacity

The call capacity is not like identifying the capacity of data traffic because of the characteristics of VoIP and video conferencing as discussed in section 2.2. The time sensitive nature of the data means that if the data does not arrive within the recommended time parameters then the quality of the call decreases. This means that call capacity is identified when the characteristics of VoIP and video conferencing fall below the recommended guidelines stated in diagram 14 and 15, which are provided by Cisco (2004a).

VoIP	
Parameter name	**Recommended Value**
Delay (End-to-End Delay)	Not exceed 150ms (one way)
Delay variation (Jitter)	Not to exceed 30ms
Packet loss	Not to exceed 1 percent

Diagram 14. Recommended values for VoIP to identify call quality

Video Conferencing	
Parameter name	**Recommended Value**
Delay (End-to-End Delay)	Not exceed 150ms (one way)
Delay variation (Jitter)	Not to exceed 30ms
Packet loss	Not to exceed 1 percent
Band fluctuation	20%

Diagram 15. Recommended values for video conferencing to identify call quality

2.7.2 Call Capacity of VoIP in WAP, MANET & ESS Environments

This section will identify from previous research the call capacity of VoIP and video conferencing over the different topologies.

VoIP over WAP

WAP is a topology that has had the most research into the call capacity of VoIP. Findings conclude that the call capacity when implemented using 802.11b is seven calls. This was from a variety of sources including Intel 2006 (Intel Cooperation 2006) whitepaper, Cisco's book entitled 'Deploying Voice over Wireless LANs' (Geier 2007) and a 2004 report by Hole et al. Cisco however specifically identifies that depending on the codec used the call capacity can be up to eight calls if the G.729 codec is implemented instead of the G.711 (Cisco Systems Inc. 2004b) due to its higher bandwidth consumption thus providing an improved quality of call. (Durkin 2003) When the call capacity has been reached the 2006 paper by David and Keegan identified that the introduction of background traffic of up to 256byte packets did not reduce the call capacity nor did it reduce the quality of the calls. In the implementation of traffic of 512byte packets it did however affect the quality of the call, with only 60% of the AP delay times being below 10ms.

VoIP over MANET

Similar to ESS there is very limited research into call capacity of VoIP in a MANET environment. This again like ESS (discussed below) is likely to be the result of its dynamic nature and the list of characteristics that can change in the topology, such as the number of neighbour nodes and node position within the environment. We cannot therefore specify a capacity in regards to any size of MANET topology as there is no research to support this. Houle et al (2006) paper concluded from their research that Ad Hoc was not currently ready for VoIP to be deployed on it. They advice that misbehaviour of the MAC layer through false detection of hidden node problem (discussed in section 2.1.2) causing routes to be declared as down and thus resulting in new roots having to be established, which subsequently adds delay.

VoIP over ESS

Research into the call capacity in ESS is very limited and could be due to the ESS environment ranging from two WAPs to hundreds of WAPs being deployed to cover a large area. As a result there is no predefined layout or limitations to provide a test bed environment. The only study that documents the capacity of ESS is a 2009 study, which identified that calls that had to be sent over two WAPs to reach there destination dropped the call capacity by 25% compared to the eight calls made over a single hop. Increasing the number of hops that calls have to cross to reach the destination caller continues to decrease capacity by three hops having a capacity of four calls, four hops two calls are supported and for five hops only a single hop can be supported. (El-Hennawey et al 2009)

Video conferencing

The capacity of video conferencing over these three topologies is very limited with only two papers that we are aware of that provides this information. The Ceo et al (2005) claim that when static nodes are used in an Ad Hoc topology that video conferencing to eight nodes was possible however when a ninth node was introduced it led to a sharp drop in performance. The second paper (Coyle et al 2006) identified that a single WAP can only support ten nodes which receive video conferencing streaming. Coyle et al experiment was providing football footage to customers' PDAs within the Ross-Ade football stadium. The capacity WAP provides for video conferencing cannot be concluded from this because in their experiment they were streaming video conferencing traffic not calling so as a result the traffic flow was heavily one way in direction from AP to client.

2.7.3 Conclusion

This section has shown that considerable studies have been documented on the call capacity of WAP in regards to VoIP from several sources identifying seven simultaneous calls being possible. Research into video conferencing over the different wireless topology is limited and was inconclusive to defining call capacity. The introduction of background traffic as identified by Davis and Keegan research (2006) demonstrated that the medium is not only limited to VoIP when call capacity has been reached. The low delay demands on VoIP and video conferencing as stated in diagram 5 and 6 above means that the call capacity can be acknowledged from identifying when performance of calls fall below those values.

3. Methods

The purpose of this section is to provide an introduction into the primary research methods used to achieve the aim of this report. The section will look at why the chosen research method was considered to be the most appropriate, along with specific details as to how the research will be carried out and the stages required to complete the research.

3.1 Primary Research Methodology and Reasoning

The primary research of this report will be obtained through an experimental based evaluation method. This will be achieved through the use of the network simulation software package called OPNET. This has been chosen as it is not only simulation software that is accessible to run this experiment on, but also as documented in Cavin et al (2002) that several popular simulators perform differently. It acknowledged that 'mismatching of the modelisation' and level of detail and configurations causes various results of each simulator to be produced. As a result the test data produced will be influenced by the software simulation package, which means that to replicate the test environment then OPNET will have to be used as the simulation software.

Through the use of a simulation package provides a number of benefits over real world testbed implementation. Three of which are identified in Almeroth et al (2003) report identifying repeatable scenarios, isolation of parameters and exploration of variety of metrics as the benefits. Additional to these identified benefits are the financial and resource limitations of projects of this scale making it unrealistic to create 'real world' environments potentially consisting of hundreds of nodes. This is supported by a significant number of researchers like Andreadis et al (2006) and Almeroth et al (2003) which within this literature review identified there primary research was carried out from network simulation.

3.2 Identification and Design of the Evaluation Simulation

This section provides details of the design and execution of gathering primary data through network simulation methodology and how it was derived from the literature review. The rest of this section is therefore broken down as follows and supported by configuration settings from appendix 3 to 13:

- Environment
- Node mobility
- Background traffic
- Data gathered
- Execution

3.2.1 Environment

Many previous simulation studies have focused on the topologies changes rather than the environment they are implemented in. (McNett, Vowlker 2005) In this report it is important that the environment is realistic to reflect the 'real world' in order for the simulation to be meaningful as identified in Cavin et al (2002) as being important. This would mean that obstacles would be incorporated into the environment that directly impact on node mobility and wireless transmissions. However a limitation of this project is the ability to create such environments due to software restrictions. Despite this the environments will be created as accurately as possible in regards to physical size and node density. Below are details of the environment that each wireless topology will be implemented within.

WAP

As documented in section 2.1.3, WAP is implemented in small environments due to its single AP deployment. In this case the environment will represent a shop floor in which VoIP and video conferencing would be used as the communication method for staff. The dimension of the environment is 80 by 60 feet with the WAP located in the centre of the environment 10 feet from the ground representing a ceiling mounted positioned WAP. The mobile nodes will have an altitude of 5.5 feet from the ground to represent an average height of a person holding a phone.

MANET

The nature of MANET means that it is suited to an environment where there is no infrastructure, as it was first developed and deployed by the military (Brouwer et al. 2007). This report will simulate the environment of an outdoor festival, where communication between people can be difficult due to the location lacking network signal coverage. The environment will not be representative of a festival of the size of Glastonbury, which supports tens of thousands of users; but instead a smaller scale festival will be represented with an environment size of 1500 by 800 feet. The mobile nodes will have an altitude of 5.5 feet from the ground to represent an average height of person holding a phone.

ESS

The literature review in section 2.2 identified that ESS is commonly deployed in enterprises and campuses, as documented in Kuang and Williamson (2005) and Henderson et al (2008). However an enterprise environment was chosen in this report as Cisco provides the recommended position of WAPs, as illustrated in diagram 16 below. The size of the area is 275 by 159 feet with APs located 10 feet from the ground, spaced 40 to 70 feet apart from each other as recommend by Cisco (2008), thus requiring a total of 22 WAP’s. There will be one smaller environment which will require less WAP’s sized at 160 x 90 feet requiring 6 WAP’s. The two different sized ESS environments will allow a conclusion to be made if the number of WAP’s and the scale of the environment have an effect on the number of calls able to be made. OSFP routing protocol will be implemented in both ESS environments since it is a non vendor specific protocol that a vast number of devices support. The mobile nodes will have an altitude of 5.5 feet from the ground to represent an average height of a person holding a phone.

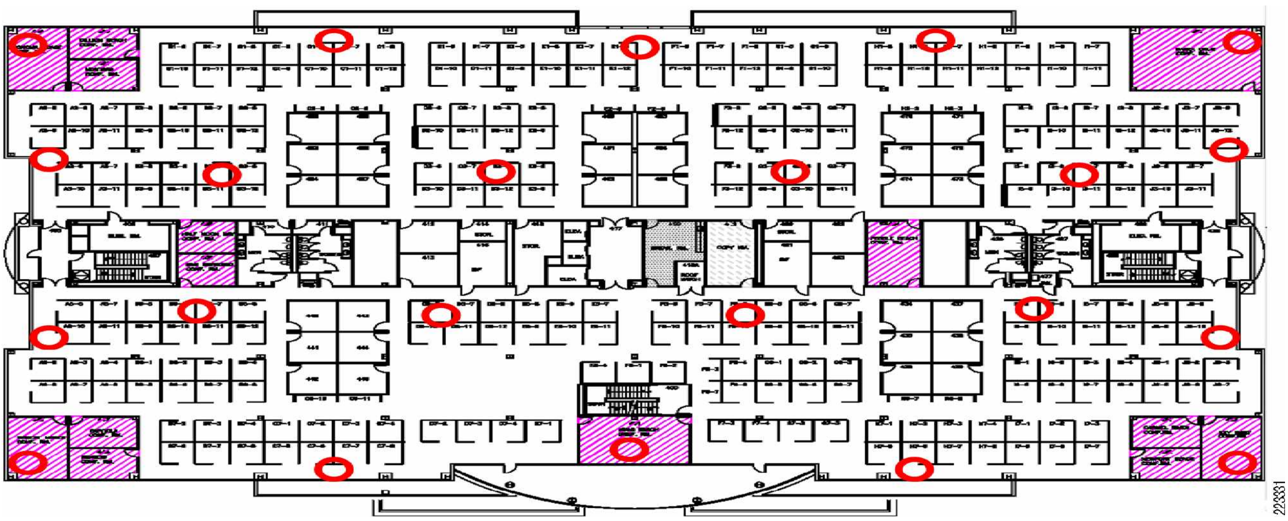


Diagram 16. Floor plan view of an enterprise building and recommended AP locations (Cisco Systems Inc. 2008)

3.2.2 Node Mobility & Population

As discussed the mobility of users within the simulation environment is one that has been described as ‘aggressive’ (McNett, Vowlker 2005) and does not reflect users’ movement who walk for a specific paths to reach their destination location (Almeroth et al 2003). Often studies into node mobility are concerned with routing protocol analysis, thus they are more interested in the topology changes than reflection of user movement within the environment (McNett, Vowlker 2005). In this report however creating accurate user movement is important as it aims to reflect user movement patterns that would be expected from users within the environment. This aims to produce more accurate results on the call capacity of both VoIP and video conferencing within each environment. The population of nodes within the environment tries reflect the number of users that would be expected to be within it. Identified by Cisco that in WAP and ESS topologies there should be no more than 15-20 nodes per AP (Cisco Systems Inc. 2004b), but due to the dynamic and random movement of users within the environment this cannot be guaranteed. In regards to MANET dynamic topology the node density cannot be predicted thus having an effect on the hop count to reach the destination node, which can have both a positive and negative impact on performance shown by data throughput to hop count in diagram 3.

The number of nodes and movement pattern (assigned randomly to each mobile node) for each environment is listed below.

WAP

The WAP environment aims to represent a shop floor, as previously noted, and in relation to its scale 20 members of staff will be represented by 20 nodes in the simulation environment. The movement of users will represent staff retrieving products for customers, updating stock and general movement within the shop. This movement will be achieved by ‘random waypoint’ movement simulation, of which three movement categorises will be created: low, medium and high; values documented in diagram 17 below. Random waypoint provides mobility to a node by randomly generating ‘x’ and ‘y’ axis values between the minimum and maximum limits of the mobility category the node is associated with. Once the node has moved the assigned ‘x’ and ‘y’ values it pauses for a specified period of time before new ‘x’ and ‘y’ values are assigned to the node and it then moves according to the new values. The problem addressed by Camp et al (2002) is that initial distribution of nodes can lead to high density areas, and this will be addressed by using a technique that involves letting the simulation be ‘warmed up’ (Noble et al 2003). As documented in Noble et al 2003 report they recognized that data should not be gathered until it enters a ‘steady state’, which in this case is once the nodes have started to move away from there initial position. The speed in which the users will move within the environment will be one meter per second as suggested in McNett and Vowlker (2005) study.

	Low	**Medium**	**High**
Minimum ‘x’ (feet)	3	32	49
Maximum ‘x’ (feet)	32	49	78
Minimum ‘y’ (feet)	3	32	49
Maximum ‘y’ (feet)	32	49	58
Pause Time (seconds)	15	25	30

Diagram 17. Values for Random waypoint categorises for WAP environment

MANET

To represent the movement of users at festivals there will be a group of a 100 static nodes positioned at co-ordinate position 0,400 to represent fans at the front of the stage barrier. There will be a further 100 nodes that will move using ‘random waypoint’ movement simulation, which the three category values for low, medium and high can be found in diagram 18. A ‘warm up’ period for the previously discussed reasons will be implemented, with node movements representative of fans walking about the festival grounds stopping to view stalls and other attractions. The speed again will be one meter per second derived from McNett and Vowlker (2005) study because it is a realistic speed user’s move at within this environment. As documented in section 2.6 the node density has a critical factor on the performance of the network in a MANET environment so there will be additional scenario that will have 300 mobile nodes in addition to the 100 static nodes located a co-ordinate position 0,400.

	Low	**Medium**	**High**
Minimum ‘x’ (feet)	50	400	500
Maximum ‘x’ (feet)	328	800	1300
Minimum ‘y’ (feet)	100	150	500
Maximum ‘y’ (feet)	300	500	700
Pause Time (seconds)	15	25	30

Diagram 18. Values for Random waypoint categorises for MANET environment

ESS

The office environment that ESS topology represents will mean that user movement will reflect walking to the photocopier, going to the toilet and seeing other staff members for example. ‘Random waypoint’ movement will be used due to its move and pause reflecting user movement within the business environment. The values for the low, medium and high mobility categorises are in diagram 19 and 20 reflecting the differentiation in the size of the environment. The number of users within both environments will consist of 80 nodes, moving at one meter per second as it is unlikely that users will be moving any faster within a work environment.

	Low	**Medium**	**High**
Minimum ‘x’ (feet)	10	50	150
Maximum ‘x’ (feet)	50	150	270
Minimum ‘y’ (feet)	10	30	100
Maximum ‘y’ (feet)	30	100	150
Pause Time (seconds)	15	25	30

Diagram 19. Values for Random waypoint categorises for ESS environment, sized 275 x 159 feet

	Low	**Medium**	**High**
Minimum ‘x’ (feet)	10	50	100
Maximum ‘x’ (feet)	50	100	150
Minimum ‘y’ (feet)	10	30	60
Maximum ‘y’ (feet)	30	60	85
Pause Time (seconds)	15	25	30

Diagram 20. Values for Random waypoint categorises for ESS environment, sized 160 x 90 feet

3.2.3 Background Traffic

Within each environment a single change of characteristic will result in a new scenario being created. In each scenario only a single change in characteristics will be made to allow accurate tracking of how changes affect the test data results. There are three characteristics that will be changed in all three wireless environments and are discussed below.

Characteristics of background traffic

The characteristics that made up background are: packet size, packets per second and traffic type. The values assigned to these three characteristics ultimately define a traffic flow. In this project three categories of traffic flows will be implemented: low, medium and high as used in a similar study by Debnath et al (2006) study. Injecting these traffic flows will therefore produce test data that can then be analysed to identify if Debnath et al (2006) findings causes call capacity to reduce when high characteristic traffic is implemented. Thus will therefore identify if their results are specific to their testbed environment or there results can be recreated in a generic test environment. The characteristics assigned to each category are listed in diagram 21 below.

	Low	**Medium**	**High**
Packet Size	128 (bytes)	256 (bytes)	512 (bytes)
Packet Per Second	10	15	20
Traffic Type	Best Effort (TCP) (http port)	Best Effort (TCP) (http port)	Best Effort (TCP) (http port)
Traffic Generated Per second	1280 (bytes)	3840(bytes)	10240 (bytes)

Diagram 21. Characteristics values for low, medium and high traffic flows

Number of nodes generating background traffic

The number of nodes that will implement the background traffic will be 25%, 50%, 75% and 100% of the node population in the environment. Using an increase scale of 25% means that when comparing results it will allow for a narrow differentiation window in performance yet scalable to large population of nodes environments such as MANET consisting of 400 nodes.

Number of calls being made between nodes

Within each wireless environment the call increase in each scenario is indicated in diagram 22 below.

	WAP	**MANET**	**ESS**
Call increase	1	10	5

Diagram 22. Call increase for each wireless environment

Through changing these variable characteristics will ultimately lead to the call capacity under different background traffic loads being found for VoIP and video conferencing calls.

3.2.4 Data Gathered

As stated in the report question delay, jitter and packet loss are the metrics that will be captured to identify when the call capacity within each environment has been reached. These metrics identify if the call is of a quality that would be expectable to the user, with their associated recommended parameter values as discussed in section 2.7. In each environment scenario the call capacity will be identified when calls and their associated gathered values are above the recommendations.

Additional data will be gathered in each environment to assist in obtaining a better understanding of the potential variation in call capacity in each environment under different background traffic demands. Diagram 23 below, identifies the different statistics that will be gathered in each environment and reasons for capturing them.

Statistic	Unit of Measurement	Environment Captured	Reason
Throughput (Per Access Category)	Bits/second	WAP,MANET, ESS	The total successfully received or forwarded bits per second for each access category. It can be used when comparing scenarios how the introduction of more background traffic and calls affects the throughput rate.
AODV - Route Discovery Time	Seconds	MANET	Represents the time to discover a route to a specific destination by all nodes in the network. By using this data along with the 'Number of Hops per Route' statistic means that data should reflect each other; with large hop per route having a longer discovery time compared to a short hop per route.
Queue Size	Packets	WAP, ESS	Records the packet queue size for each category of traffic through the packet ToS field on the wireless interfaces of the WAP router. This will aid in identifying how increasing traffic flows affects a router's queues sizes.
Data Dropped (Buffer Overflow)	Bits/second	WAP, ESS	For each access category of traffic that a WAP handles, this statistic will identify how many bits per second are dropped due to buffer overflow, which we predict will closely relate to the queue size statistic.

Diagram 23. Additional data statistics gathered in each environment.

3.2.5 Execution

Time

The execution of the simulations will consist of each scenario running for a simulation duration of seven minutes. The seven minutes is broken down as follows:

- 0-2 minutes: Allows for simulation ‘warm up’ as discussed previously.
- 2-6 minutes: Background traffic starts for the percentage of nodes that the current scenario implements.
- 3-5 minutes: The calling party initiating the call to a random node within the environment, with the call lasting 2 minute, which is greater than identified in McNett and Vowlker study (2005) which concluded in there university testbed that medium length VoIP calls lasted 41 seconds. However as the objective of this report is to identify call capacity, longer call duration is being used to provide clearer test data to analyse .Despite this it is still a realistic period of time for a phone call to last.
- 5-7 minutes: This is the period of time after the calls have ended to make data analysis of the calls easier to view as the start and end of the call duration will be easily indentified rather than ending the simulation at the end of the call duration. It also means that data can be captured for any data that is delayed when being sent through the network.

Number of Runs

Each scenario will be run twenty times, which should provide a large enough sample size to take into account the randomness of node mobility and reduce the probability of a ‘fluke’ result in regards to call capacity being identified. As discussed on MathWorks website (MathWorks 2010) it is important to change the seed number of each run so that events associated with the generated random number will not be generated for the twenty runs. Thus for each of the twenty runs for all scenario the seed value will start at a value of five and be incremented by five so the twentieth run will have a seed of hundred.

Implemented Wireless Technology

The implemented wireless technologies that are to all nodes and WAPs all testbed environments is the 802.11e standard, which provides QoS required for time sensitive traffic as discussed in section 2.4. The physical layer wireless technology which operates below the 802.11e standard on the nodes and APs will be the 802.11g standard, which was published in 2003 (IEEE 2009). The transmission power of the node devices have been reduced in each environment to reflect the size of the environment which also in the real world devices reduces battery consumption. Within the WAP environment the transmission power is 5E-006 providing a range of 40 feet, MANET transmission power is 6.5E-005 providing a range of 150 feet and in both ESS environments the transmission power is 6E-006 providing a range of 46 feet.

In the MANET environment, AODV will be implemented as the routing protocol as it is an ‘on demand’ protocol reducing control traffic generated but also as discussed in section 2.5, nodes do not need to know routes to all other nodes in the environment. Added to this, it has been documented that AODV perform within 2-4 % of DSR and DSDV, which are table-driven routing protocols (McNett, Vowlker 2005).

Implemented VoIP Codec

The voice codec that will be used in VoIP calls is the G.711 standard as it was identified to be more tolerant to longer delay and acceptable call quality when compared to other voice codec's (Durkin 2003). Edo. et al (2009) further backups Durkin points by identifying that due to G.711 having no compression it provides a call quality like that of a ISDN phone network since it implements the same G.711 codex. Additional to the call quality that no compression offers, it also aims to provide the lowest time latency since less compression is required to be processed at either end of the call (Edo et al 2009). The characteristics of the voice call are shown in diagram 24 below.

Characteristic	**Value**
Bits per Second	120,000
Packet Size	120 (bytes)
Packet Per Second	250
Traffic Type	UDP

Diagram 24. Traffic characteristic for G.711 voice codec

Implemented Video Conferencing Codec

The codec implemented for video conferencing is H.323 as documented by Ried (1999) which documents its implementation over IP networks and wide deployment in enterprises since its release in 1996. A low quality video conferencing call will be implemented due to the assumption that users using a mobile device are unlikely to require a high quality call like that which a dedicated conference room can provide. The traffic characteristics that the video conference call has are shown in diagram 25 below.

Characteristic	**Value**
Bits per Second	138,240
Frame Size	17280 (bytes)
Frames Per Second	10
Traffic Type	UDP

Diagram 25. Traffic characteristic for H.323 low quality video conferencing call

What is noticeable when comparing the characteristics of VoIP and video conferencing is the increase in 'Bits per Second', this a direct result of the additional media stream of video. The H.323 codec implements the G.711 codec to handle the audio part of the call, so it can be identified in this implementation that an additional 18240 bits per second are sent in order to support the video stream.

4. Presentation of Results

This section of the report will evaluate and document the findings of the primary data gathered from the execution of the network simulation as documented in the methods section 3. The call capacity of VoIP and video conferencing within each wireless topology will be identified and aims to evaluate how the introduction of different characteristic background traffic flows effects the call capacity and performance.

4.1 WAP: VoIP Calls

4.1.1 Sent and Received Traffic with no Background Flows

In section 2.7.2, previous studies and documentation indicate that the call capacity of a single WAP environment in optimal condition is 7-8 calls. They implemented the 802.11b standard, which only supports 11 Mbit/s compared to the implemented 802.11g that supports 54 Mbit/s in this report experiment. Diagram 26 below, illustrates the average voice packet traffic sent and received for 8, 9 and 10 VoIP calls placed in a WAP environment with no background traffic flows, for a call period of 120 seconds. Thus from the environment not containing any background traffic flows it can be described as the 'perfect network environment' conditions since there is no competing traffic flows except other calls wanting access to the WAP.

As documented in section 2.7.1 the ITU recommendation for packet loss is 1% in order for the call to be acceptable to the user. It is clear that when 10 calls are implemented this recommendation is not met for the duration of the 120 seconds that the call lasts. It is only during the time period of 265 and 275 seconds in which the traffic sent and received is within the 1% and approximately 125 packets are being received by the 10 calls before received packet levels drop. From the data gathered from the 'Buffer Overflow' of the WAP no packets were dropped by any of the calls including the 10 calls thus it can be assumed that lost packets are due to the nature of wireless technology as discussed previous in section 2.1.2 and 2.6. The performance of 8 and 9 calls clearly indicates the negative impact adding a tenth call being placed on the network has, but shows that a single WAP with no background flows can in regards to call traffic sent and received meet the ITU recommendation.

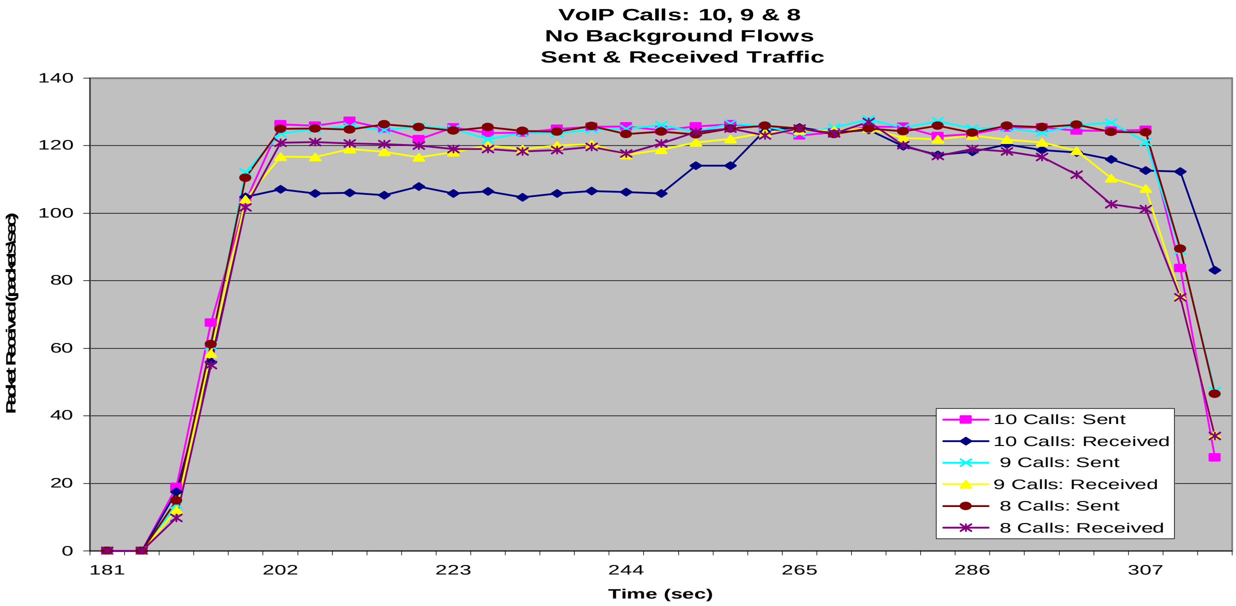
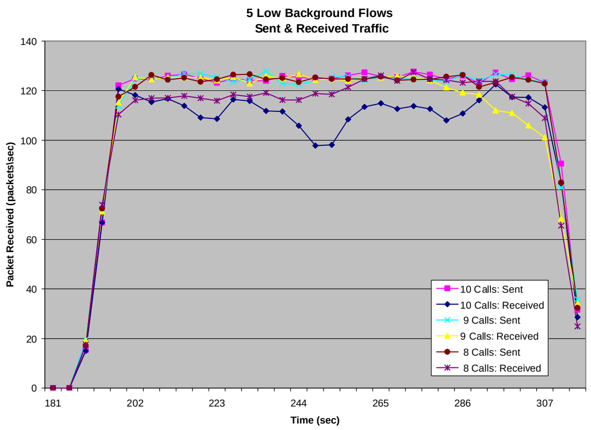


Diagram 26. Average traffic sent and received for 10, 9 and 8 VoIP calls with no background traffic flows

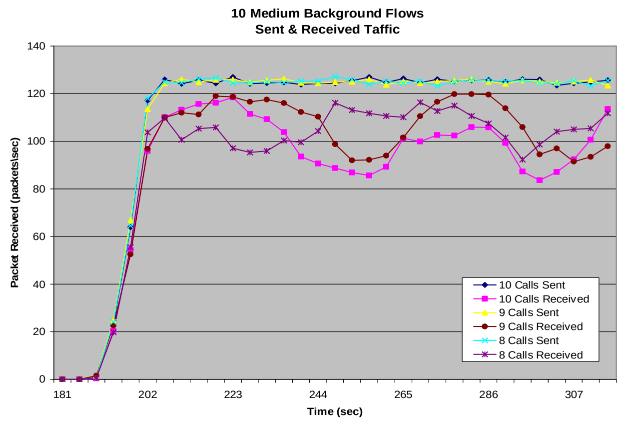
4.1.2 Sent and Received Traffic with Background Flows

Five nodes implementing low background traffic flows is illustrated in diagram 27 (i) below and is the lowest traffic demand in regards to number of flows and traffic demand characteristic tested. This further concludes that 10 calls is not possible in the implemented WAP environment and supports the documented literature. It is noticeable is that 9 calls out performed 8 calls, which would not be expected due to the additional demands that an additional call places on the network. However with mobility and the nature of wireless medium as discussed in section 2, this again could be the reason for this result. Additional to this if the number of runs for each scenario was increased to above 20 then the expected results may have been achieved due to a larger sample size, this is further discussed in section 6.6.

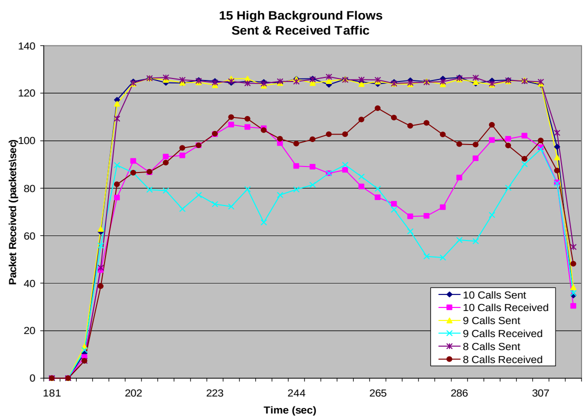
The introduction of 10 nodes sending medium background traffic flows shown in diagram 27(ii), demonstrates that the characteristics increase in packet size and packets per second of the background traffic flows does have an impact on the average call quality in regards to voice packets received. It is noticeable that all three call scenarios in 27(ii) were affected by this increase in traffic demands, most noticeable by 10 calls is the significant large drop in received packets during 240 and 260 seconds time period. In the case of implementing 15 high background flows, diagram 27(iii), which is 75 % of the nodes population within the environment generating high background traffic flows, there was a visible sharp drop by the performance in the 9 calls category. At the time period of 286 seconds in 27(iii) there is a drop of 65 packets to that being sent by the 9 calling nodes and compared to the same time point under medium traffic (diagram 27 (ii)) there is a difference of 60 packets.



(i)



(ii)



(iii)

Diagram 27. Average sent and received traffic for 10, 9 and 8 VoIP calls with: 5 low background traffic flows (i), 10 medium background flows (ii) and 15 high background flows (iii)

4.1.3 Further Analysis into 10 VoIP Calls

As discussed above WAP is unable to provide the QoS needed in regards to packet delivery reliability in order for the user not to experience a call that would have short periods of silence due to dropped data. Another aspect of the QoS required for VoIP and a measurement of the call quality is End-to-End (ETE) packet delay, which is shown below in diagram 28. It illustrates the average ETE delay of voice for 10 calls under different background traffic scenarios for the 120 seconds call time, which starts at 180 seconds. During the 120 second call time it is noticeable that the calls on average ETE delay are clearly below the recommended 150ms except for a fluctuation during 190 to 208 seconds for scenario '10 Low BG Flows'. The result of this fluctuation in ETE delay is the result of the buffer queue size reaching 90 packets as illustrated in diagram 29 due to peak VoIP traffic being reached at this period of time. As a result, diagram 30 highlights the number of packets lost during this time (190 to 208 seconds) being almost the equivalent to the that of 10 high background flows, which generates 89600 bytes a second more traffic compared to 10 low background flows. The cause of the fluctuation can thus be concluded due to the low background flow packets being smaller in packet size therefore more time is required by the WAP to send the same number of bytes as 10 medium and high background flows. The QoS mechanism then revaluates its scheduling algorithm and priorities the VoIP traffic, which during this time, the queue size drops significantly (diagram 29) as a result of the buffer dropping packets (diagram 30).

As identified in section 4.1.2, WAP is unable to provide the QoS of 10 VoIP calls under perfect environment conditions that of no background traffic for the period of time between 260 and 270 seconds. Diagram 30 illustrates this point further, showing the significant drop in bits due to buffer overflow and is why the queue sizes remain relatively similar under all conditions, as demonstrated in diagram 29. Despite the number of bits being dropped appearing high it is however a relatively few packets due to a single G.711 call producing 120,000 bits a second. Thus as would be expected under 20 high background flows the buffer peeked at approximately 296 seconds at 750,000 bits which can be viewed as 6 individual call packets being dropped at that precise second in time.

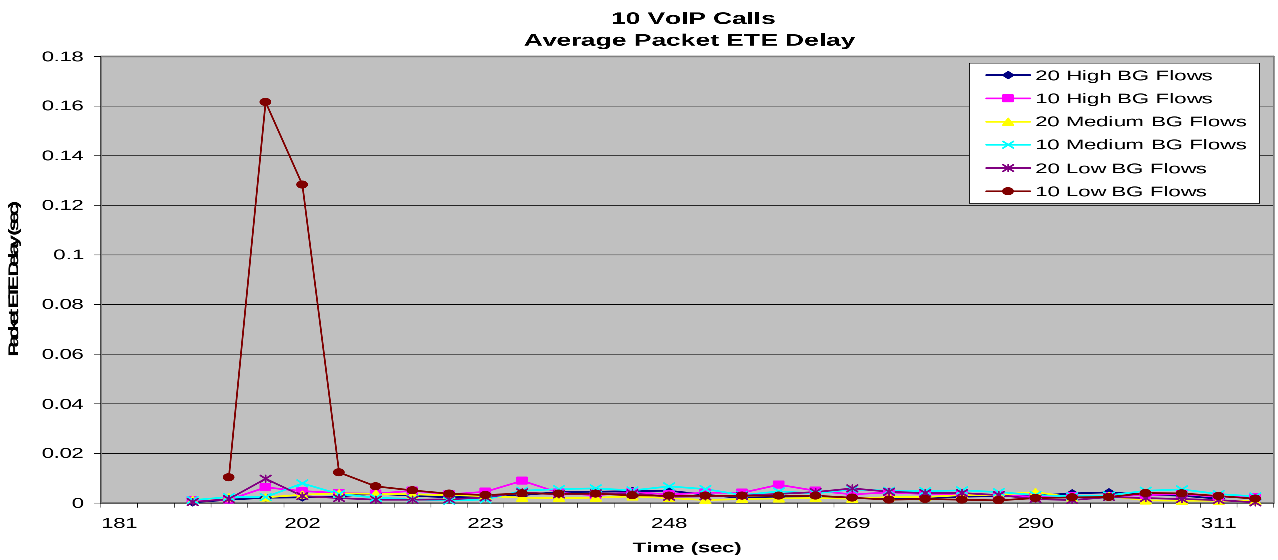


Diagram 28. Average packet ETE of voice traffic for 10 VoIP calls under different background flows

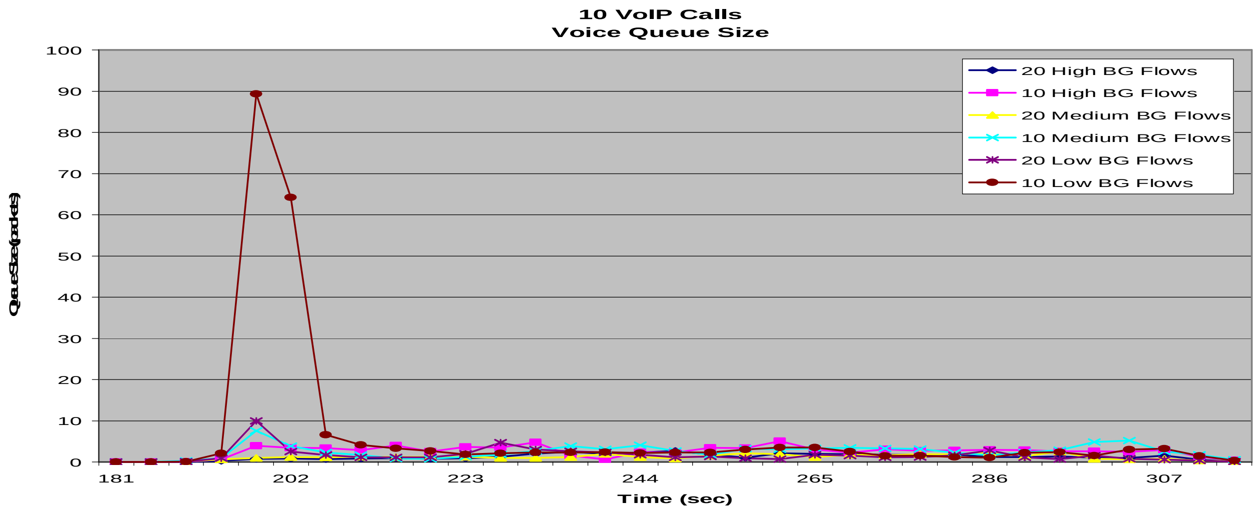


Diagram 29. WAP voice queue size for 10 VoIP calls under different background flows

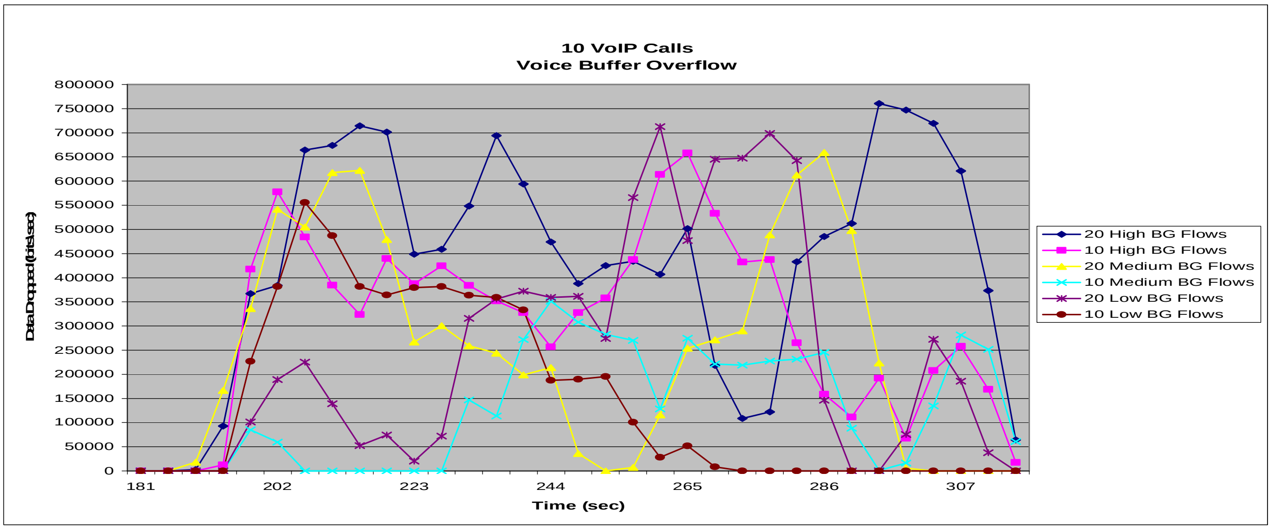


Diagram 30. WAP voice buffer overflow for 10 VoIP calls under different background flows

4.1.4 Further Analysis into 9 VoIP Calls

As identified that 9 calls were able to be reliably delivered the average ETE delay for 9 calls under varying background flows is shown in diagram 31 below. It is noticeable that on average under all background flows the ETE delay remains around 0.0004 seconds, which is significantly lower than the recommended limit of 150miliseconds. Under high background flows it can be noted to perform as well as lower background flows and this is reflected by the voice queue size (diagram 32) under high background flows remaining relatively low, below 0.001 of a packet. When 9 calls queue size is compared to that of 10 calls it can be seen that there is a significant drop from the average of 1 to 4 packets to 0.00075 packets in the case of 9 calls. This low queue size is not the result of packets being dropped as under all background flows scenarios implementing 9 calls no bits were dropped due to buffer overflow. Again this further demonstrates the impact of a 10^{th} call has on the queuing buffer and therefore on network performance.

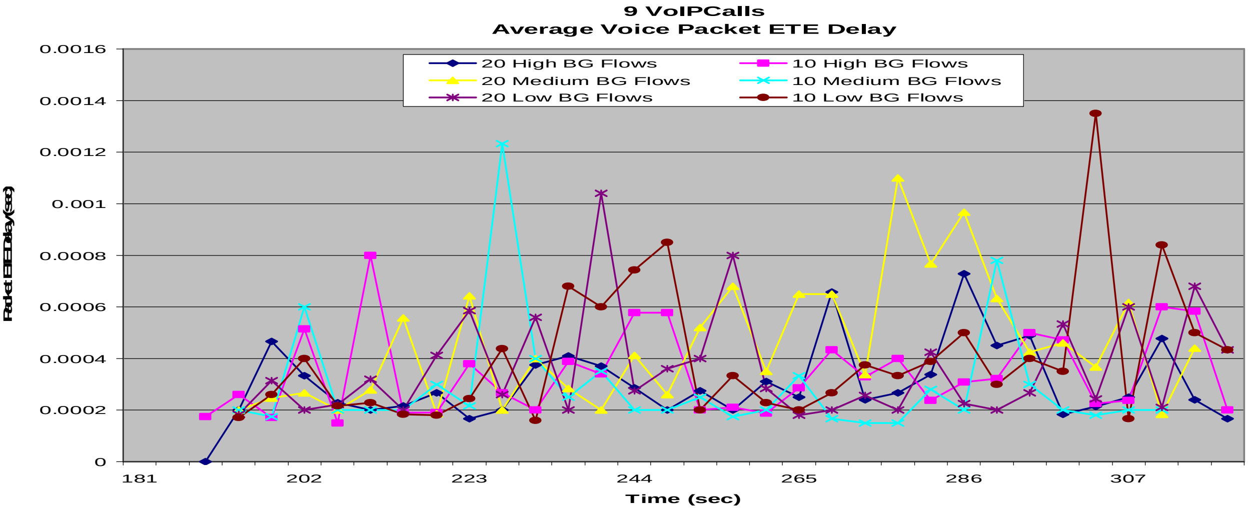


Diagram 31. Average packet ETE of voice traffic for 9 VoIP calls under different background flows

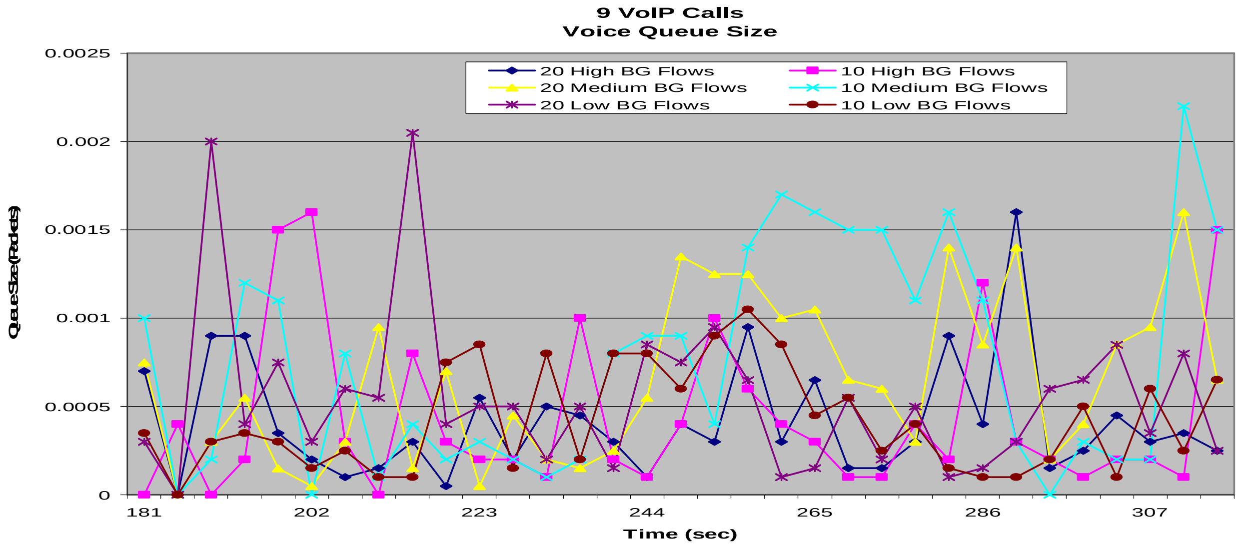


Diagram 32. WAP voice queue size for 9 VoIP calls under different background flows

4.1.5 Further Analysis into 8 VoIP Calls

Diagram 33 below, shows the ETE end of the voice traffic for 8 calls. When it is compared to the 9 calls ETE delay (diagram 31) it is noticeable that their performance is relatively similar with the average value being between 0.0002 and 0.0006 seconds. Despite there being no bit drop under all 8 calls scenarios due to buffer overflow, in the case of 20 high background flows the ETE delay performance is similar to that of 9 calls, in which it remains considerably consistent for the duration of the call at 0.0002 seconds. In relation to voice queue size these similarities continue when comparing the performance between 8 and 9 calls, demonstrated in diagram 34 below for 8 calls and diagram 32 above for 9 calls.

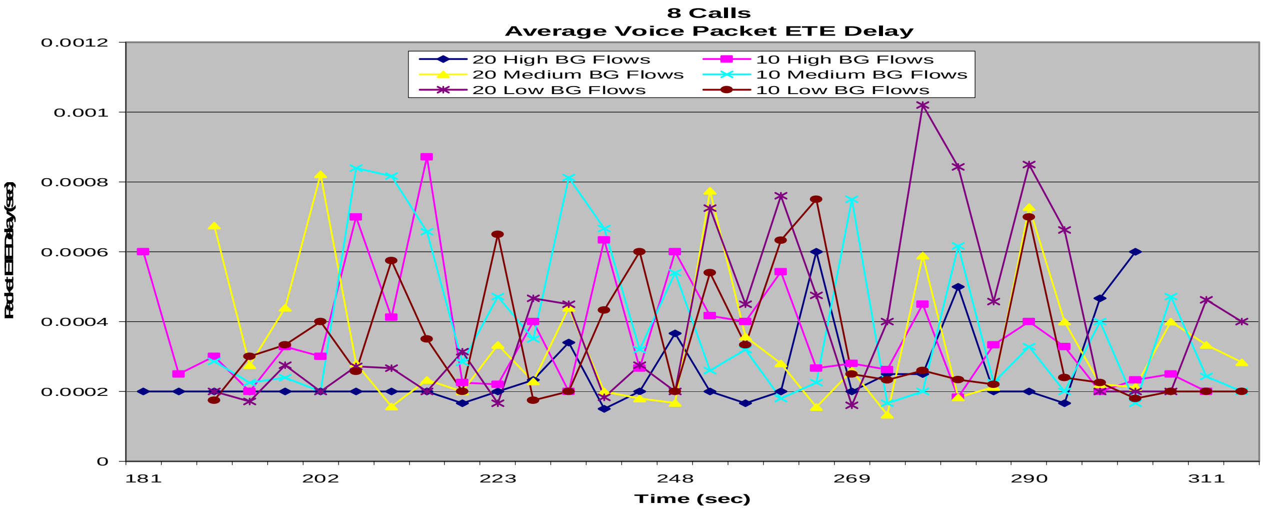


Diagram 33. Average packet ETE of voice traffic for 8 VoIP calls under different background flows

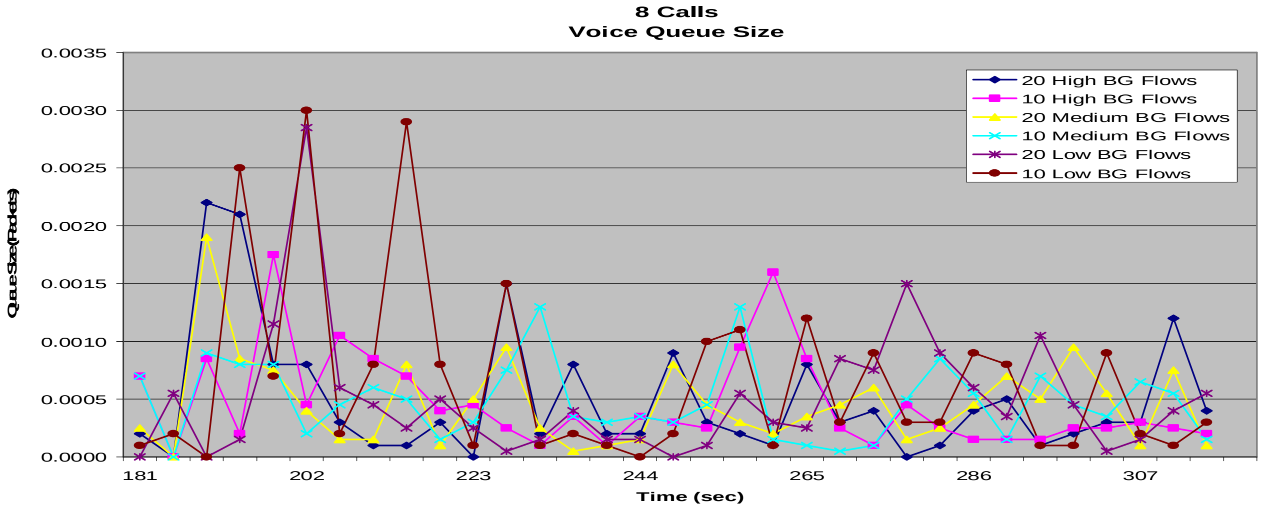


Diagram 34. WAP voice queue size for 8 VoIP calls under different background flows

4.1.6 Conclusion

As stated from previous literature research that implemented 802.11b for WAP their call capacity was 7-8 calls. From this research it can be concluded from the primary data gathered that 9 calls is possible when implementing the 802.11g standard, with the addition of a 10^{th} call having a significant negative impact on the network performance. The addition of the 10^{th} call results in the call quality dropping below the recommended ITU recommendation as demonstrated from the received packet loss, shown in diagram 24 above. In addition to 9 calls being able to be made, it can also be concluded that even under the heaviest traffic condition of 20 high background flows, there was no significant impact on the voice calls ETE delay that made it not meet the recommended ITU standards. This differs from David and Keegan (2006) study previously discussed in section 2.7.1 that concluded that implementing packets of 512bytes significantly affect the delay of the calls. The ability to cope with these large background traffic flows can be said to be due to the 802.11g standard supporting and is implemented in this report allowing a bandwidth of 54 Mbit/s, thus able to deliver more bits per second and therefore reducing queue size compared to the David and Keegan 802.11b standard implementation. However when the average successful received voice packet rate is looked at in regards the introduction of background traffic flows, it shows the impact on call capacity.

It can be concluded from diagram 27 that once 10 medium traffic flows are introduced onto the network that none of the calls (10, 9 and 8) were able to successfully receive voice packets that meets the ITU standard of 1% packet loss. Thus from no packets being dropped from buffer overflow in either of the 9 and 8 call category, the drop packets can be seen to be due to the nature of the wireless medium in the reliability of ensuring the node receives the packet as it moves within the environment.

5.2 WAP: Video Conferencing Calls

5.2.1 Sent and Received Traffic with no Background Flows

As documented in the literature section 2.7.2, the call capacity of video conferencing has not been identified before over the wireless medium. Diagram 35 below, illustrates the average packet sent and received for the range of 5 calls to just a single call under the 'perfect network environment' as there is no background traffic flows.

It can be clearly seen that under the range of calls tested that no calls were able to obtain the successful transmission of packets to provide a call quality of only 1% packet loss. At the time 265 to 270 seconds for '1 call' the packets sent was equal to that received by the receiving node however it starts to decrease sharply after this point to 6.5 packets at 286 seconds and follows a similar increase in received packets during 286 to 300 seconds. These increases in packet delivery can be shown as a result of the video queue size decreasing to below one packet as shown in diagram 38(i) and additional data from diagram 39 that the buffer drop bit decreases to almost 0 bits a second. It can be seen that once the packet queue size increases above one packet then the successful delivery of packets cannot be obtained during the 180 to 300 second call time. As would be predicted as the number of calls increase the number of successfully received packets drops, as shown in diagram 38 by '5 calls' receiving on average the lowest number of calls continuing the trend to calls 1 and 2 that receive the highest number of packets. However when this scalability is considered in regards to buffer overflow (diagram 39) there is a significant increase of more than 3 millions bits being lost between calls 2 and 3 to calls 4 and 5. This demonstrates the saturation point of the WAP has been met when the 4th call is placed on the network, with the number of bits 5 calls drops being noticeably higher than 4 calls when compared to the bit drop between calls 2 and 3.

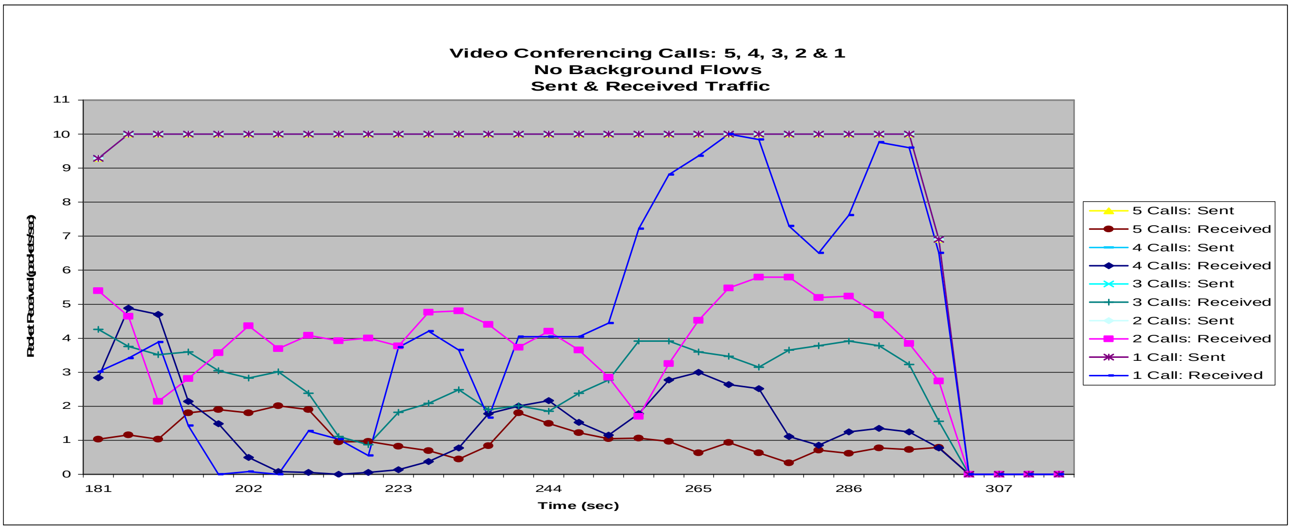
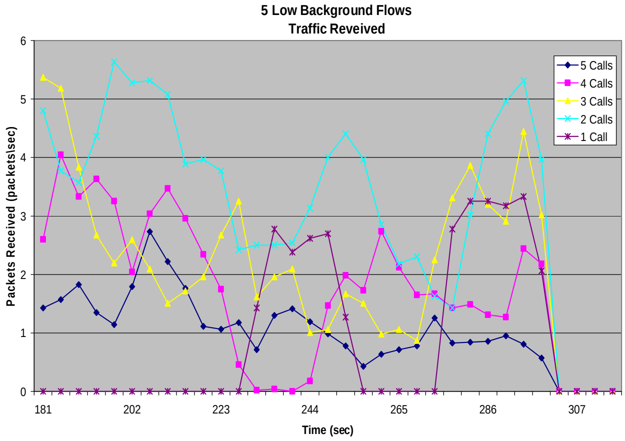


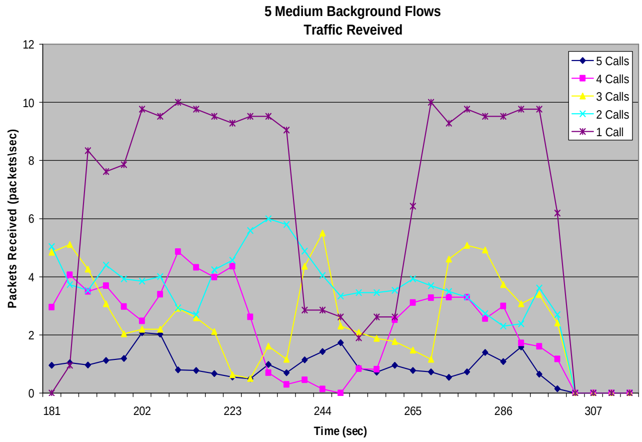
Diagram 35. Average sent and received traffic for 5, 4, 3, 2 & 1 video conferencing calls with no background traffic flows

5.2.2 Traffic Received with Background Flows

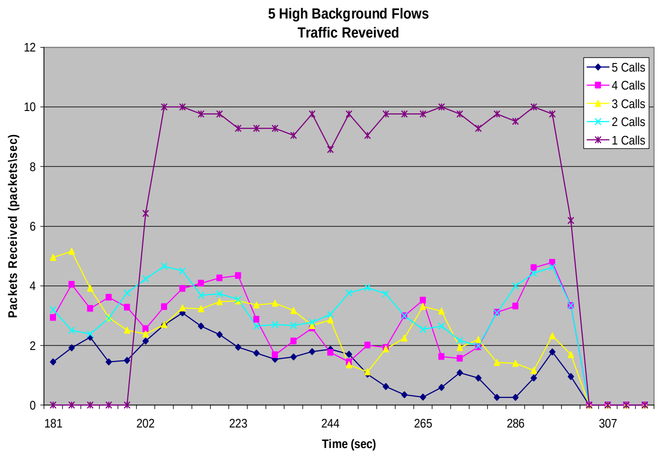
As identified in diagram 35 above, the WAP was unable to support a single video conferencing call when there was no background traffic being produced by the nodes in the environment. Diagram 36 below demonstrates and emphasises the lack of reliability of successfully being able to send and receive packets over the wireless medium. In both cases 36(ii) and 36(iii) show that despite increased traffic flows through the introduction on 5 medium and high background traffic flows respective to their diagram, was for a single call being made 8 to 10 packets were able to be received for significant periods of time. Specifically under high background traffic, which was close to achieving a call that meets the ITU recommendations of a 1% packet loss. When 5 calls are placed on the network it can be seen in 36(i), 36(ii) and 36(iii) to achieve on average the poorest average received packets however what is visible is that under all background flow types is its average received packet being approximately between 1 and 2 packets a second during the length of the call. Thus result can be seen to be obtained due to the QoS and the prioritising of video traffic over background best effort traffic.



(i)



(ii)

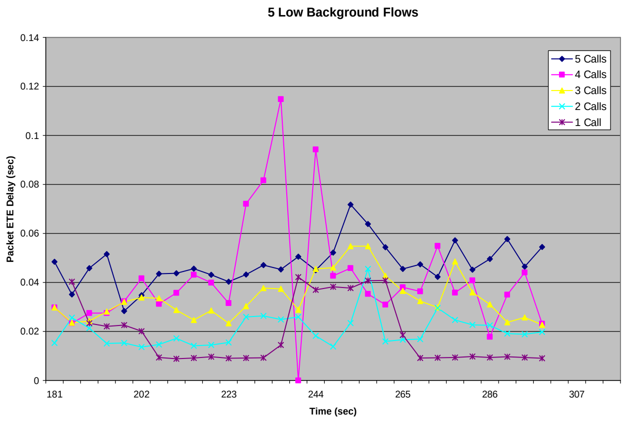


(iii)

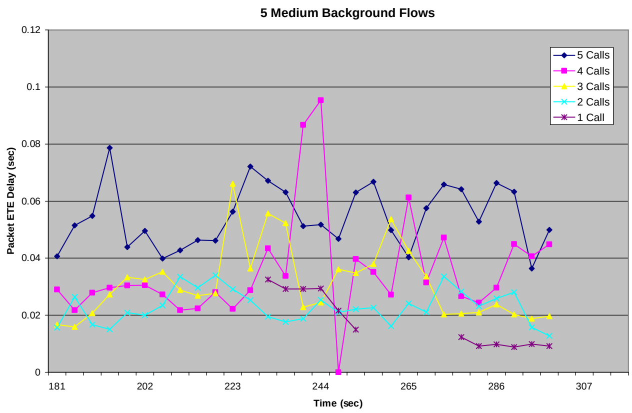
Diagram 36: Average received traffic for 5, 4, 3, 2 & 1 video conferencing calls with 5 background flows: low (i), medium (ii) and high (iii)

5.2.3 ETE Delay

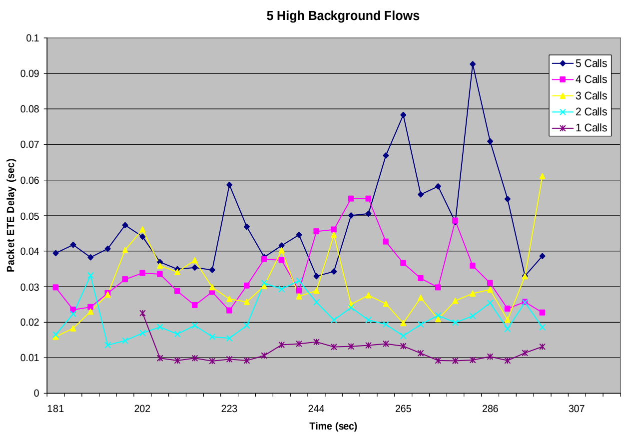
With no calls being able to achieve with or without background flows in regards to successfully receiving the expected 10 packets is of interest to this report as it provides greater insight into the ability of WAP being able to support video conferencing calls. In diagram 37 below it can be seen that the ETE delay under all call scenarios was able to achieve the ITU recommendations for the packets that were successfully received by nodes. It is noticeable and could be argued to be expected is that in all scenarios in diagram 37 below is that '5 calls' experienced the highest average ETE delay. Thus being argued the result is because more calls are trying to be sent through the WAP, in which all calls have the same priority and therefore the WAP treats them in a FIFO queuing manner (discussed section 2.3.4) and are delivered in the order they arrive at the WAP.



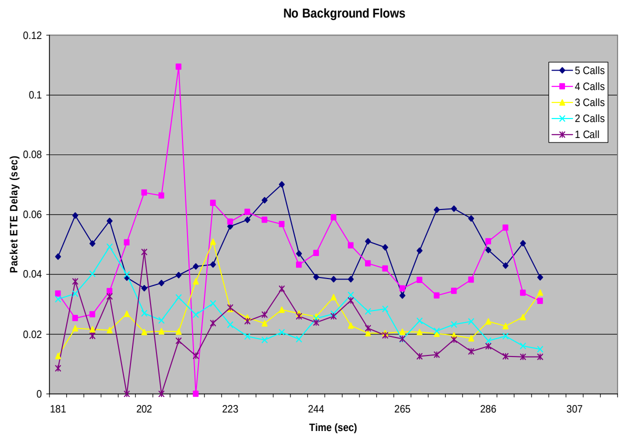
(i)



(ii)



(iii)

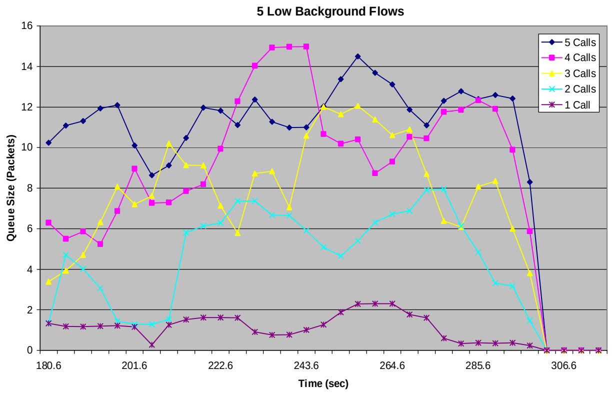


(iiii)

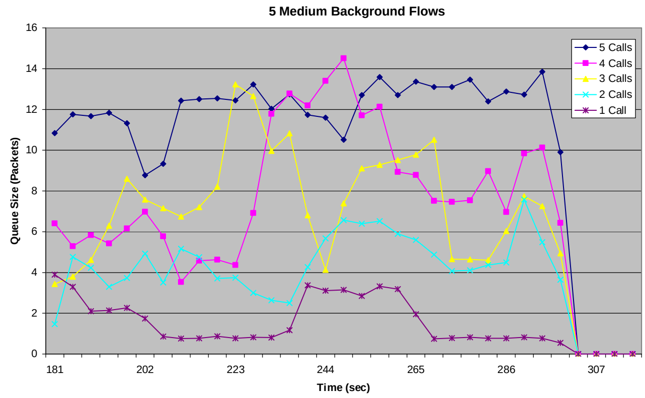
Diagram 37: Average ETE delay for 5, 4, 3, 2 & 1 video conferencing calls with 5 background flows: low (i), medium (ii), high (iii) and none (iiii)

5.2.4 Video Queue Size

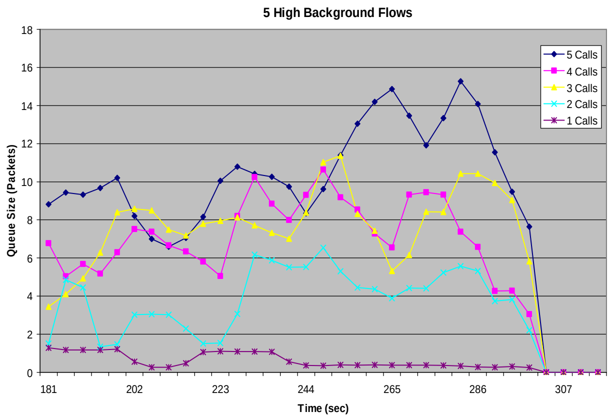
The queue size for video traffic is shown directly below in diagram 38, for calls 1 to 5 under different background flow conditions. What can seen is similar queue lengths for each call under different background traffic flows, for example ‘1 call’ remaining between approximately between 0.3 and 3 packets and ‘5 calls’ between 8 and 14 packets for the duration of the 120 second call which starts at 180 seconds. This trend shown can be explained due to the queue size for video being unaffected by the background traffic flows because it is stored in the best effort queue, thus it is the number of video calls that affect the video queue size.



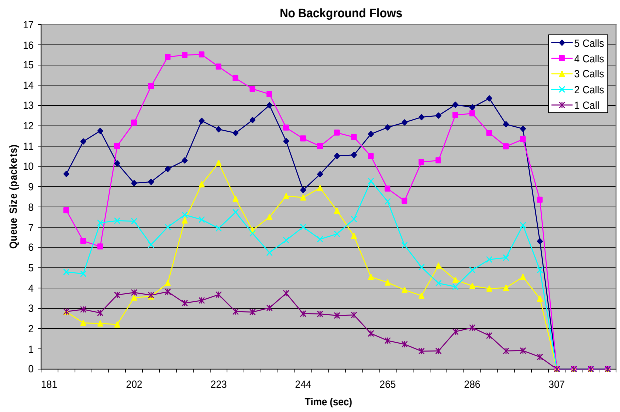
(i)



(ii)



(iii)

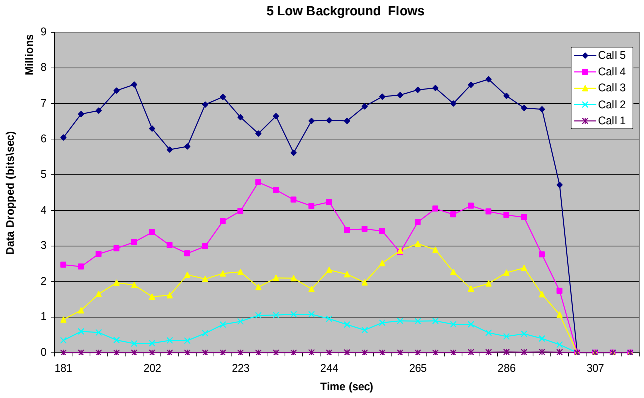


(iii)

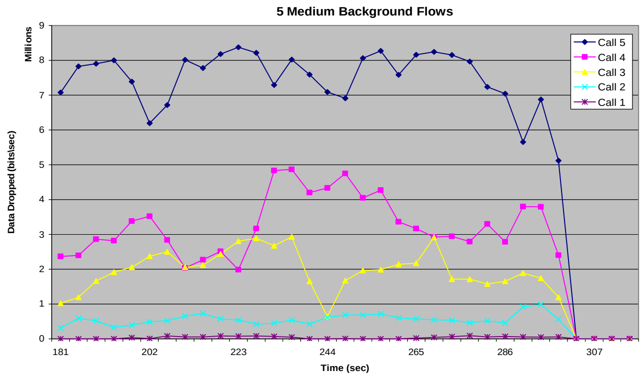
Diagram 38: WAP video queue size for 5, 4, 3, 2 & 1 video conferencing calls with 5 background flows: low (i), medium (ii), high (iii) and none (iiii)

5.2.5 Video Buffer Overflow

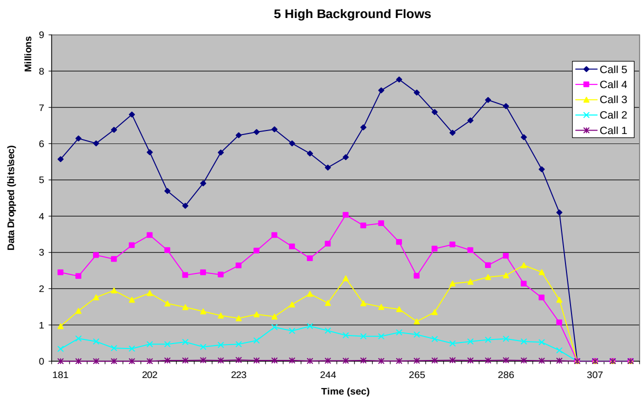
The number of bits per second dropped by the video buffer is shown in diagram 39 below. As seen by the queue size data results discussed above, the data dropped as a result of the video buffer overflowing shows that background traffic flows do not affect the video buffer performance because data in regards to background traffic flows is stored in a separate queue. What can be identified is the significant increase in dropped bits from the implementation of the 5th call compared to 4 calls. This increase can be seen to be as much as 5 million bits at the time point of 286 seconds in the medium background traffic flows scenario (diagram 39ii). The low bit drop experienced by a single call reflects the low queue size, which is shown in diagram 38 above, thus as result means that one, there is not enough video traffic being produced to cause the queue buffer to overflow and second through the QoS mechanism giving priority to sending video traffic means the queue does not build up a queue of packets thus again causing the buffer to overflow.



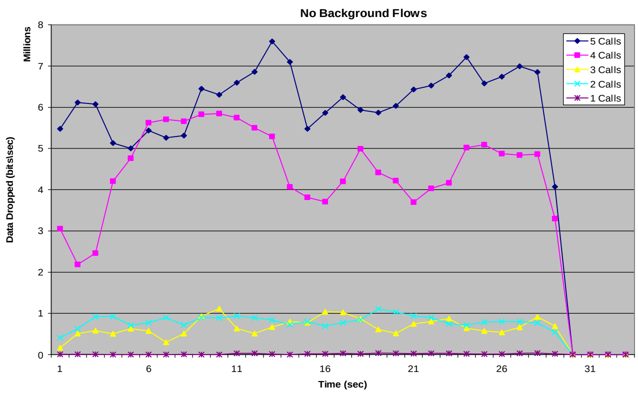
(i)



(ii)



(iii)



(iiii)

Diagram 39. WAP video buffer overflow for 5, 4, 3, 2 & 1 video conferencing calls with 5 background flows: low (i), medium (ii), high (iii) and none (iiii)

5.2.6 Conclusion

As diagram 35 and 36 clearly illustrate video conferencing calls are not possible in the tested WAP environment, with not even a single call under 'perfect network environment' being able to provide reliable received packets delivery. The performance of 9 VoIP calls which produces 135000 bytes a second, when compared to a single video conferencing call produces 17280 bytes a second. Thus VoIP requires more bytes to be sent however the significant factor is the size of packets that each type of call uses. The VoIP codec uses small packets of 120 bytes each whilst video conferencing uses frames of 17280 bytes, which even when a frame header is added to the VoIP packets there is still a vast difference in frame size. Thus as a result means that the WAP buffer is unable to store a large amount of video conferencing packets due to their size, demonstrated in diagram 39 showing that buffer overflow increases as the number of calls increases most noticeable between calls 4 and 5.

5.3 MANET 200 Nodes: VoIP Calls

The nature of MANET and its lack of infrastructure topology as documented in the literature review section 2.1.3 means that establishing the route from caller to called party will be achieved through the AODV routing protocol, discussed in section 2.5. Thus means routes to the destination have to be first established and then maintained as the nodes move within the environment, which was documented in section 2.6 and is a wildly discussed challenge to solve.

5.3.1 Traffic Received with no Background Flows

Diagram 40 below shows that of a MANET environment containing 200 nodes in regards to the average call traffic received, which identifies that a significantly less than the expected 125 packets was achieved. The nature of the MANET environment means since there are no dedicated fixed routing devices as in WAP and ESS environment makes the route the packets takes very dynamic and thus utilises different node resources dependent on the node position. This is demonstrated by how 30 calls resulted in higher average received packets compared to 20 calls. A reason why 100 calls produced such low performance in regards to voice traffic received is because half the nodes in the environment are calling, thus generating traffic and therefore either a route to the destination

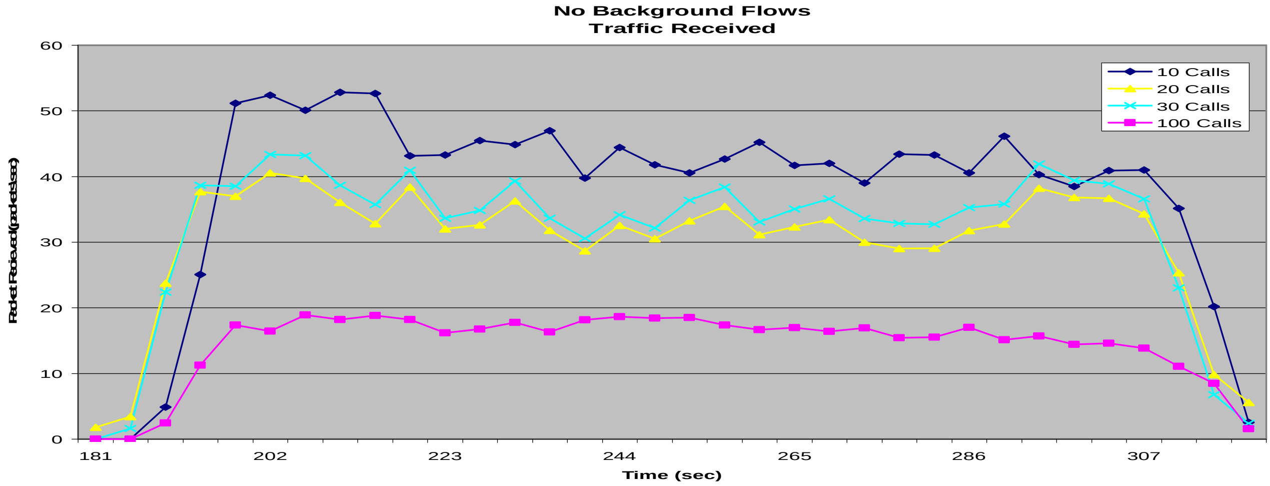


Diagram 40. Average received traffic for 10, 20, 30 and 100 VoIP calls with no background traffic flows

5.3.2 Traffic Received with Background Flows

The introduction of background flows shown in diagram 41 below, demonstrates a noticeable increase in the number of packets received by the called party when compared to the same number of calls with no background traffic flows (diagram 40). This increase can be seen due to the dynamic nature of the MANET topology by the non predetermined routes traffic is sent across the network. What is particularly noticeable in diagram 41(iii) is that during the time period of 252 and 300 seconds is that approximately 50 extra packets were received under medium and high background traffic flows. An explanation for this is documented in section 6.6.

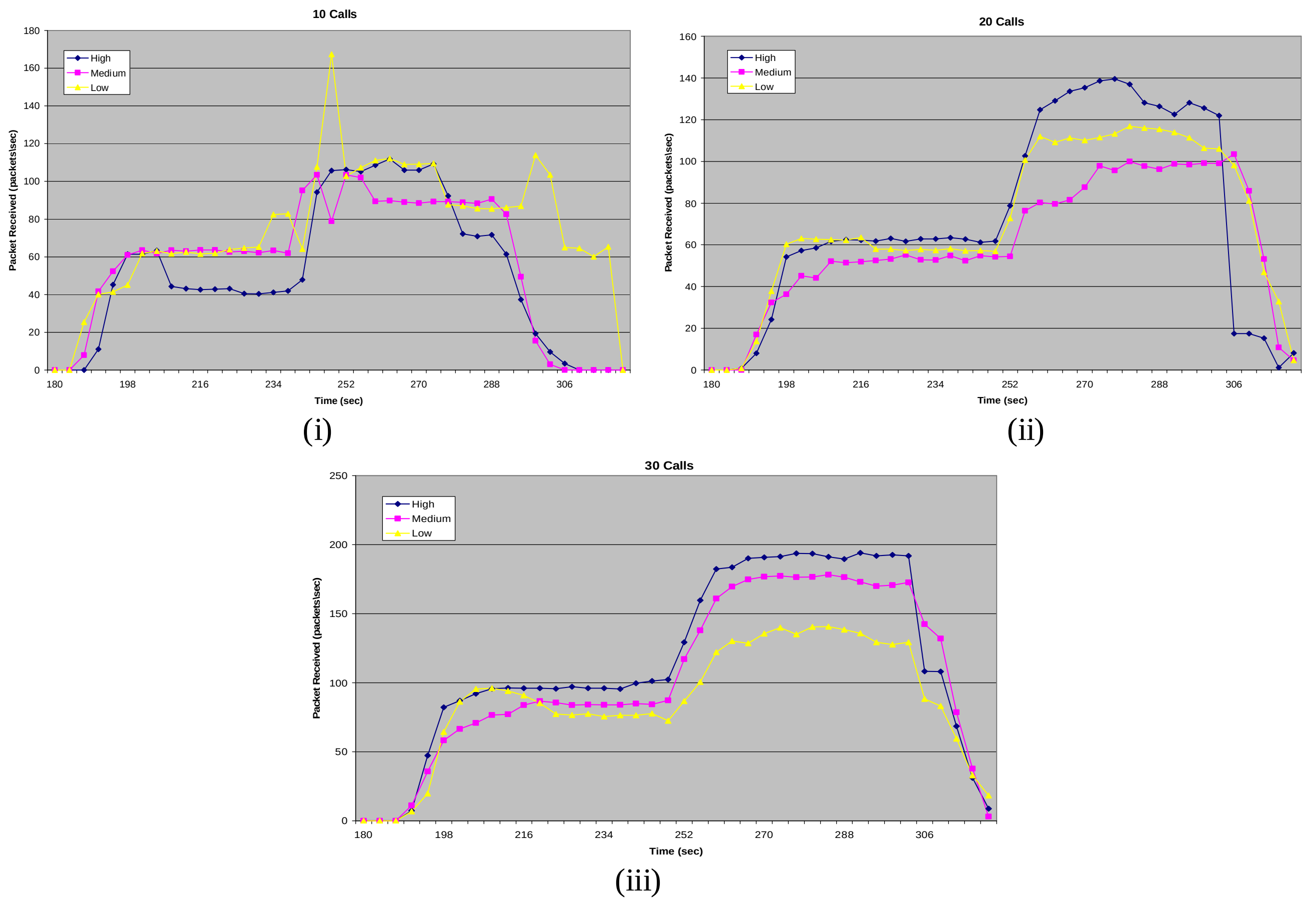


Diagram 41. Average received traffic for 100 high, medium and low background traffic flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.3.3 ETE Delay with no Background Flows

10 VoIP Calls

ETE delay is an important factor in regards to time sensitive traffic such as VoIP in this environment as for each hop the packet takes along the route path is subjected to several delays as discussed in section 2.2.2. Thus the result of high ETE delay user conversation will experience moments of silence potentially causing both users speaking at the one time. Diagram 42 below demonstrates the ETE delay for the 10 individual calls in diagram 40 for the duration of the 120 seconds the calls last, with calls starting at 180 seconds. The high ETE delay of call 2 is noticeable at both 214 and 277 seconds, which would ultimately have caused the user to end the conversation after 214 seconds. These and other ETE delays that calls such as 3, 6 and 7 experience can be said to be due to the route to the destination being lost and having to be established as shown in route discovery time in diagram 46(i) below. There are clear peaks at these time spots (214, 277 and 298 seconds) where high discovery times occur and thus directly affecting some of the calls ETE delay.

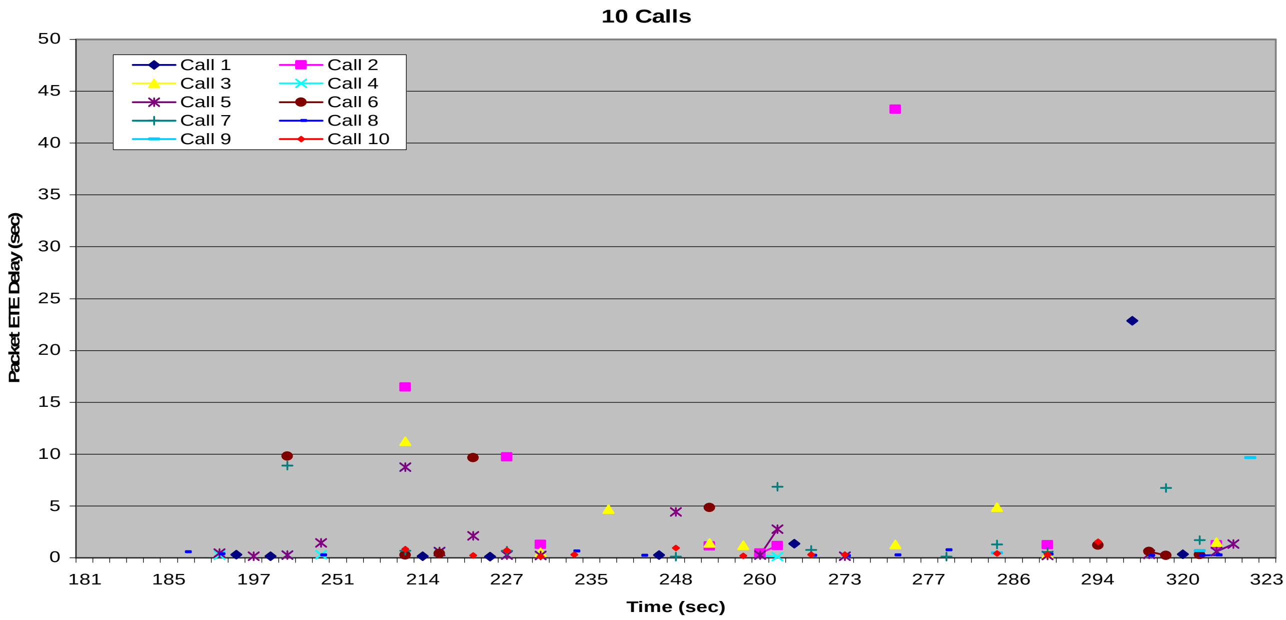


Diagram 42. Average packet ETE of 10 VoIP calls with no background flows

20 VoIP Calls

Diagram 43 below shows the ETE delay for 20 calls, which as identified for the ETE for 10 calls above, the route discovery time impacts on the ETE delay of some calls notably in this scenario for calls 2, 10 and 11. This when referenced with diagram 46(ii) identifies sharp increases in discovery times when the route to the destination had been lost.

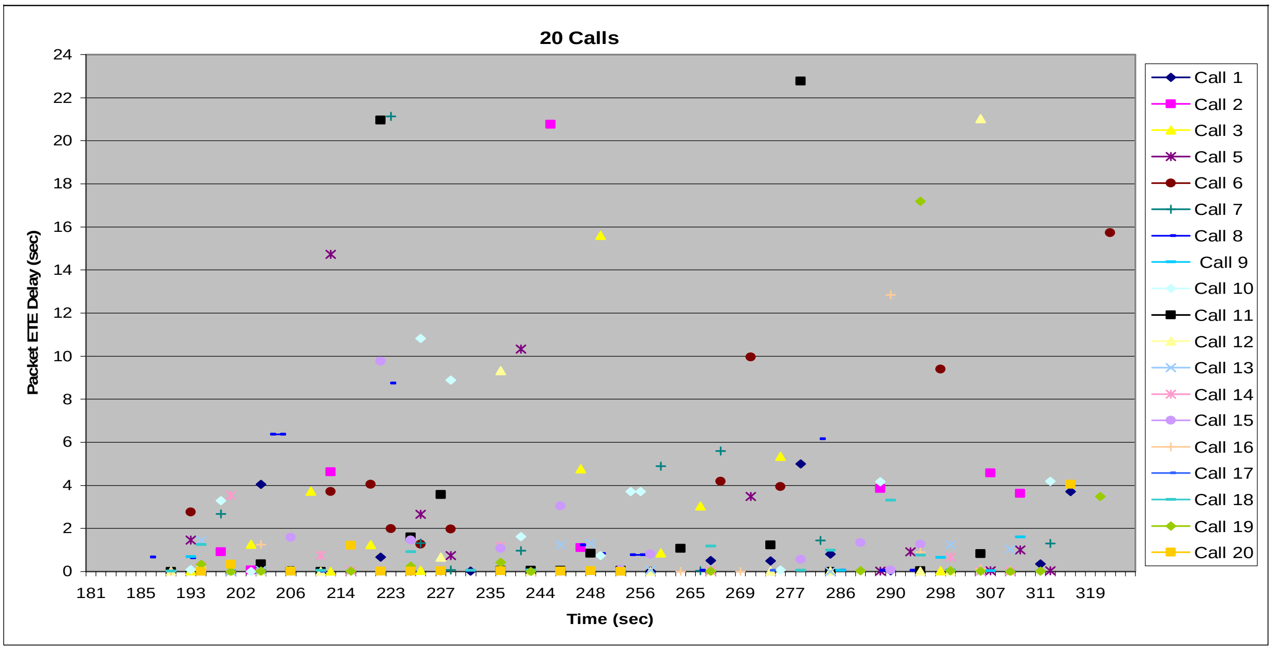


Diagram 43. Average packet ETE of 20 VoIP calls with no background flows

30 VoIP Calls

When 30 calls ETE delay is evaluated in diagram 44 below and is then referenced to sharp peak increases in discovery time in diagram 46 (iii), the information further concludes that route discovery time directly impacts on the performance of ETE packet for specific calls.

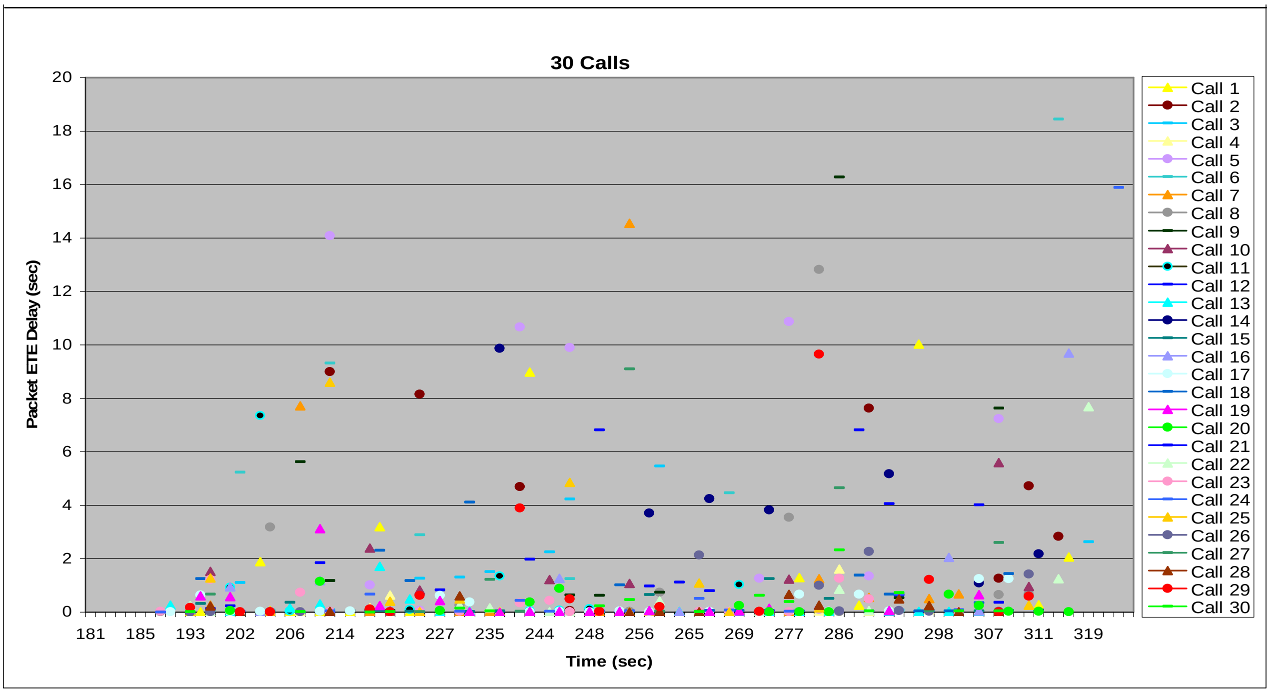
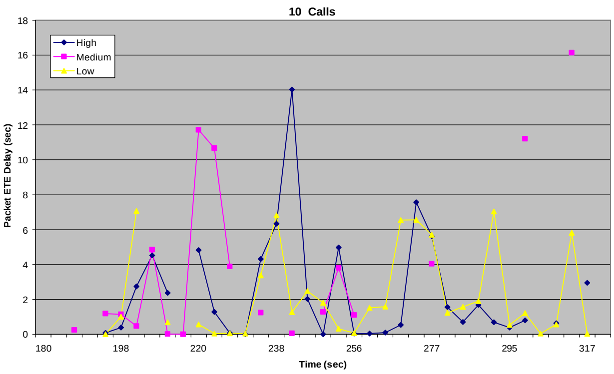


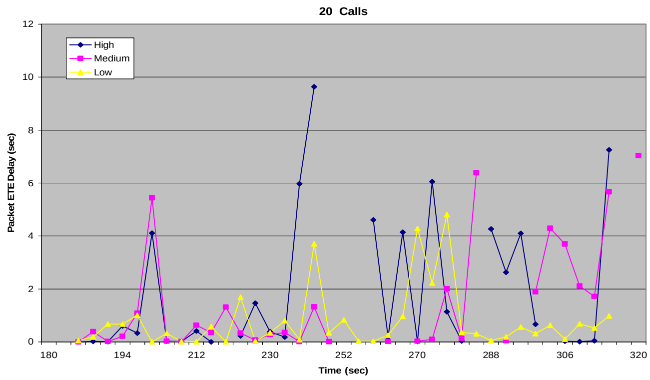
Diagram 44. Average packet ETE of 30 VoIP calls with no background flows

5.3.4 ETE Delay with Background Flows

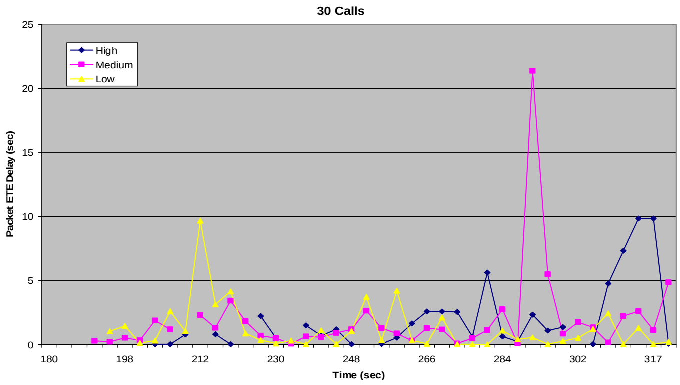
The average ETE delay for voice is shown in diagram 45 below for calls under different demands of background traffic flows. As shown by the increased number of packets received (diagram 41) there is therefore more ETE delay data being produced through more calls successfully receiving packets. What diagram 45 illustrates is that despite there being increased demands on the network the ETE delay of the calls were not noticeably affected when compared to calls made with no background flows. When 30 calls with and without background flows; diagram 40 and diagram 45 (iii) respectively, are compared it can be seen that for the duration of the call period that ETE delay remains between 0.1 and 3 seconds. Thus at points the ETE delay falls within the ITU recommend of 0.15 seconds.



(i)



(ii)

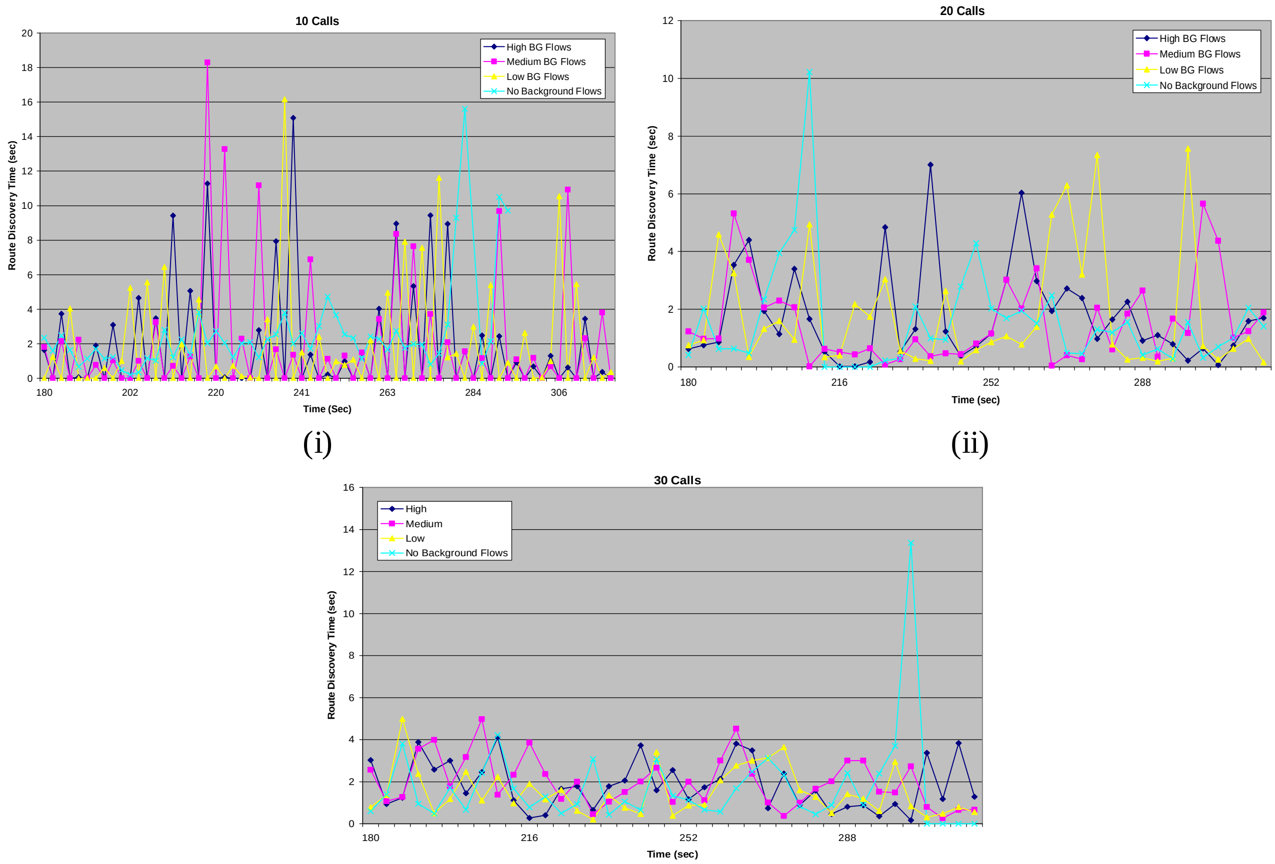


(iii)

Diagram 45. Average packet ETE for 100 high, medium and low background traffic flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.3.5 Route Discovery Time

With MANET being a non fixed infrastructure topology and running an ‘on demand’ routing algorithm (AODV) means that the route discovery time is an important aspect since it defines the length of time for a route to be established to the destination node. Thus diagram 46 below shows the length of time under different background flows and call demands affects the route recovery time, with both background flows and VoIP calls being incorporated into this time. It is noticeable in all three diagrams that despite the discovery time of background traffic being incorporated into the discovery time value there is no clear difference in performance compared to a VoIP only scenario. It can be further documented this would be expected by AODV route establishment mechanism discussed in section 2.5. This is to be expected because no data traffic is sent by the sending node until a route has been established between the source and destination nodes but also due to routing packets being marked in the ToS field in the packet header with the highest priority thus getting priority to be sent first by the nodes.



(i) (ii) (iii)

Diagram 46. 'Route Discovery Time' for 200 high, medium, low and no background traffic flows for: 10 calls (i), 20 calls (ii), 30 calls (iii)

5.3.6 Background Flow Throughput

The main focus of this project is to find out the call capacity within each environment however the performance of background traffic is also important since users may be more interested in using the network for email access or web browsing rather than making a VoIP call. Diagram 47 below, shows how the increase in background flows effects the throughput in relation to the different traffic types: low, medium and high with 10 VoIP calls being made. It is noticeable as the number of flows increase there is not a significant rise in the throughput in regards to bits per seconds; however peaks in throughput occur for example in diagram 47 (ii) at 227 to 245 seconds and 320 to 340 seconds. These peaks indicate that for the majority of simulation time background traffic is unable to find or maintain a route to their destination. Thus it can be assumed to be due to the environment testbed, a 200 node population is not sufficient to provide routes to nodes in all areas of the environment that a node can move to.

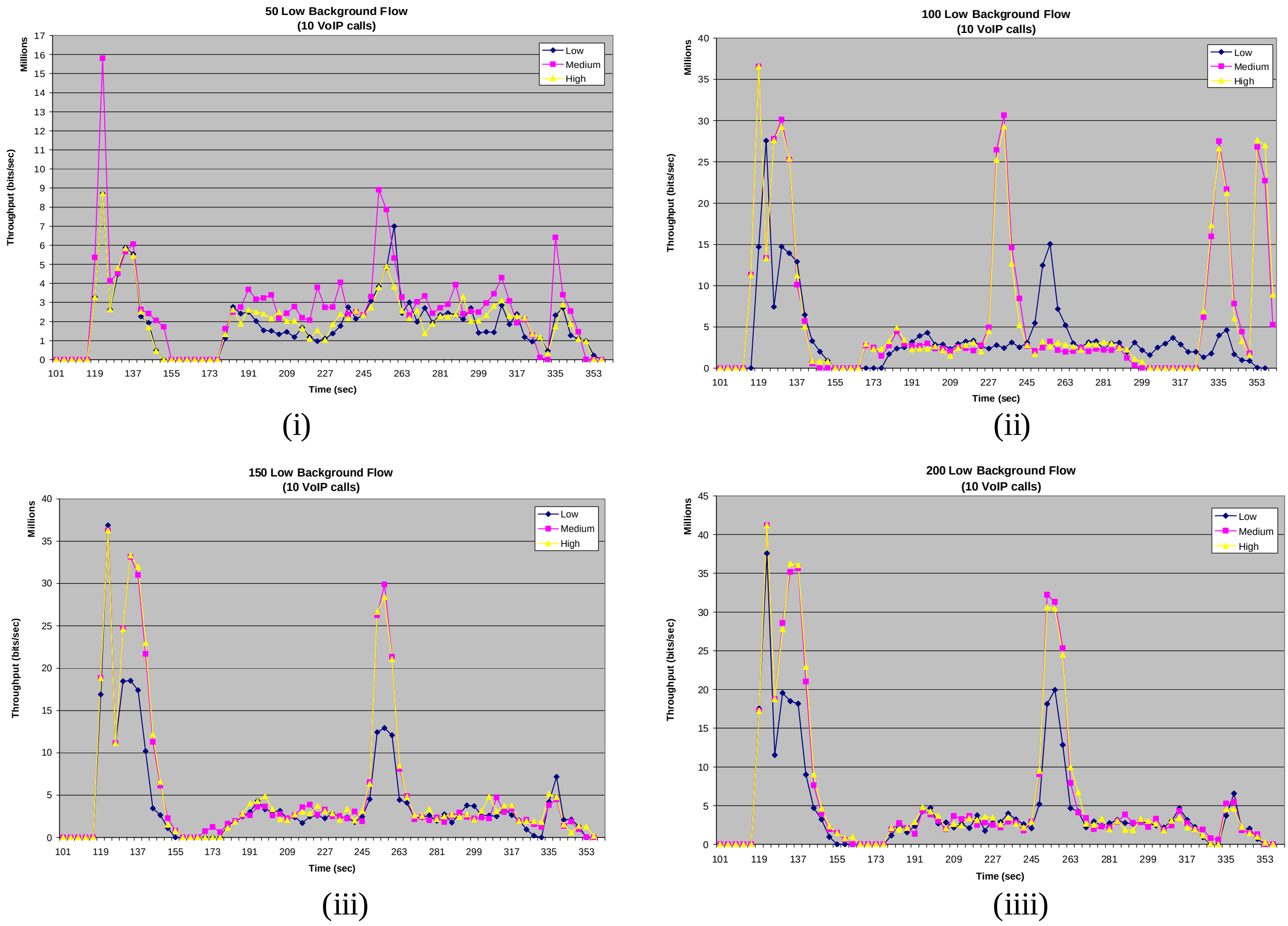


Diagram 47. Data throughput of background flows under 10 VoIP calls under: 50 BG flows (i), 100 BG flows (ii), 150 BG flows (iii) and 200 BG flows (iiii)

5.3.7 Conclusion

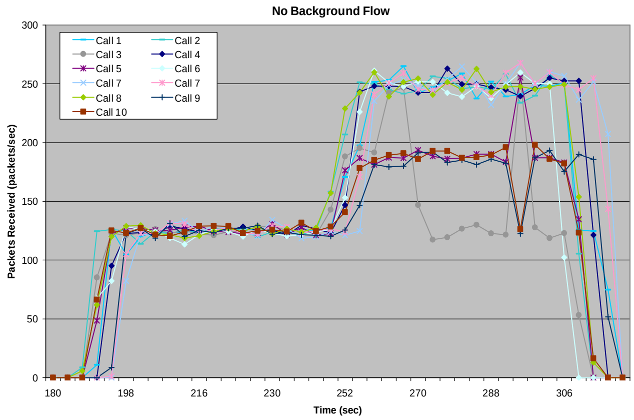
A 200 node MANET environment on average was unable to support 10 calls as shown in diagram 40, which in an environment consisting of 200 nodes would be a minimum number of calls that the network can handle whilst meeting the ITU recommendations. The long ETE delay experienced by all calls (10, 20 and 30) as referenced to diagram 45 identified that route discovery process has a direct impact on the ETE delay. As documented in section 2.6 as nodes move and break established route links then new routes have to be established and thus stopping packets from being sent to the destination until a new route can be established. The analysis of the background traffic throughput and the steep increase in throughput further aids in supporting that 200 nodes does not make the environment node dense enough to provide a high probability that a route to a destination node will be established. In regards to route discovery when implemented under different background traffic flow demands, there is no noticeable difference in performance due to AODV routing packets receiving the highest QoS metric thus its queue on node devices are served with the highest priority. Thus it is rather the position of nodes that dictate the route discovery time.

5.4 MANET 400 Nodes: VoIP Calls

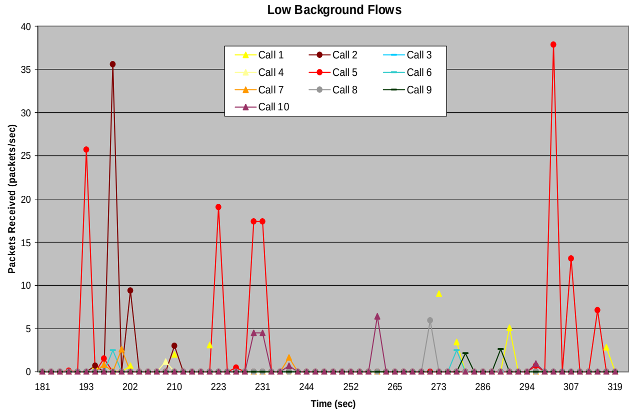
The conclusion of the 200 nodes discussed in section 5.3 was that the environment did not contain enough nodes to provide routes to destination nodes. Through the increase of 200 additional mobile nodes, discussed below is how these additional nodes affect network performance.

5.4.1 10 Calls Traffic Received

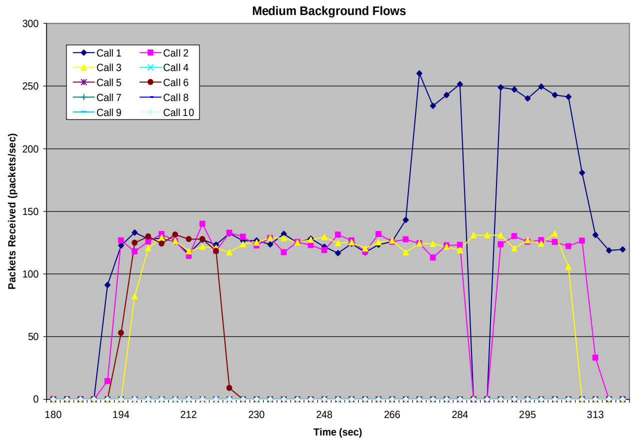
Diagram 48 below shows 4 different scenarios for the average packet received for 10 calls under no background flows and low, medium and high 400 background flows. As discussed previously the expected received packets that a VoIP node would expect to receive is 125 packets a second in order for the call to meet the ITU standard. It is noticeable from diagram 48(i) that all 10 calls successfully received 125 packets a second for the duration of the 120 second call. This is a clear increase in performance compared to 10 call implementation in a 200 node environment, thus suggesting that increasing the node count has a noticeable positive effect on network performance. Also evident and occurred under the 200 node scenario is that on average the majority of the nodes after the first minute of the call period receive double the number of packets than would be expected and is discussed in section 6.6. The introduction of background traffic into the network appears to have a negative effect on the call capacity in regards to achieving 10 calls as demonstrated in 48(ii), 48(iii) and 48(iiii), in which all see a drop in successful calls. Under both medium and high traffic flows both scenarios were able to support 2 to 3 calls for at least 60 seconds. Diagram 48(ii) however shows the lack of infrastructure and reliability of packet delivery that MANET environments support as despite generating lower packet demands compared to medium and high background demands no calls were successfully achieved.



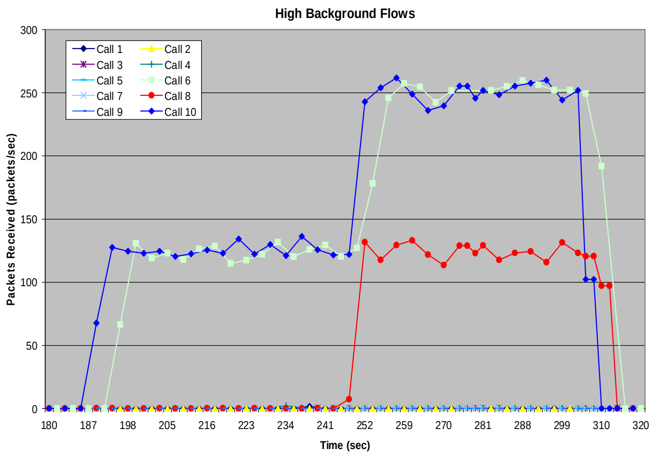
(i)



(ii)



(iii)

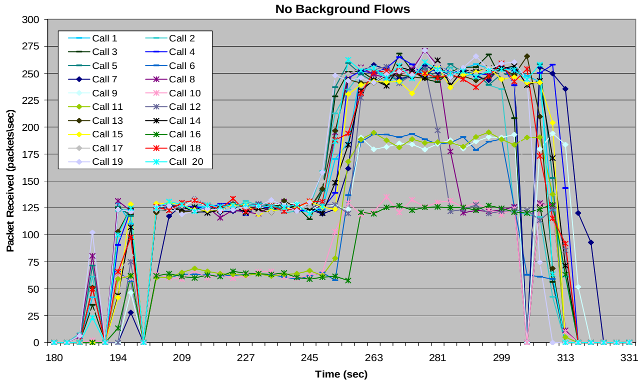


(iiii)

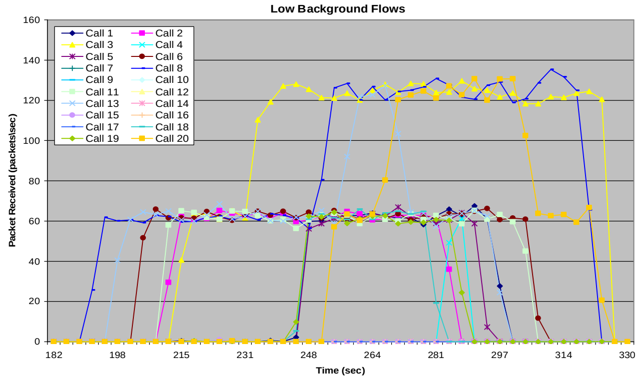
Diagram 48. Received traffic for 10 VoIP calls under: no BG flows (i), 400 low BG flows (ii), 400 medium BG flows (iii) and 400 high BG flows (iiii)

5.4.2 20 Calls Traffic Received

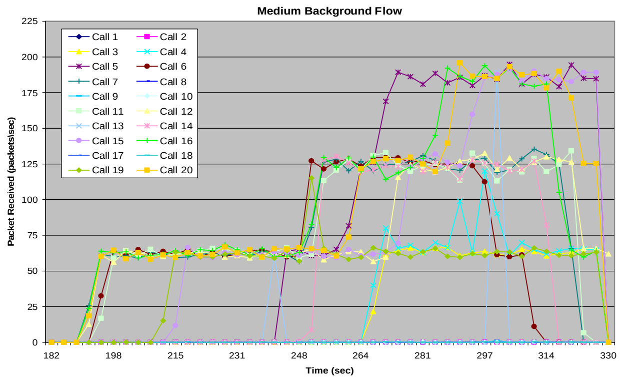
The introduction of more calls in a MANET environment due to 10 calls being successful, it cannot be assumed that by adding new calls they will be successful or even retain the 10 call capacity. MANET and its dynamic nature means that node position is a critical factor in data successfully being able to be routed to the destination node. Diagram 49 below shows the voice packets received by the receiving node for all 20 calls under different background network flows; low, medium and high. What is noticeable in all sub diagrams in diagram 49 is the receiving of the double number of packets that are being sent by the sending node and is again discussed in section 6.6. Again under no background flow demands and can be seen as the ‘perfect network environment’ shown in diagram 49(i) where all 20 calls for approximately 60 seconds during the simulation time of 245 and 300 seconds received the sent 125 packets per second. The introduction of the additional 200 nodes and the increase in performance is demonstrated through the number of calls that start receiving traffic several seconds after the expected call start time of 180 seconds. For example in 49 (iii) calls 15 and 19 do not start to 200 seconds with call 3 not receiving packets until 264 seconds, which is 84 seconds after the expected start time.



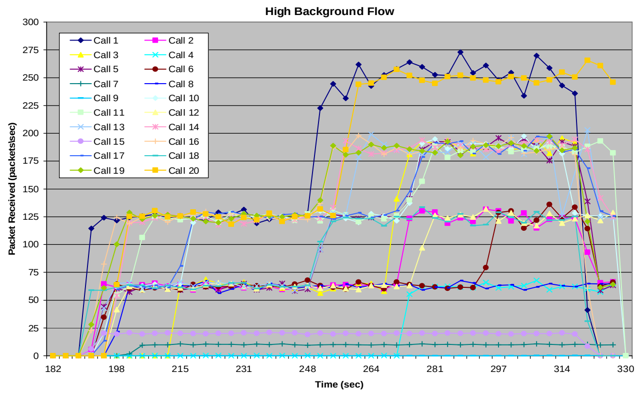
(i)



(ii)



(iii)



(iiii)

Diagram 49. Received traffic for 20 VoIP calls under: no BG flows (i), 400 low BG flows (ii), 400 medium BG flows (iii) and 400 high BG flows (iiii)

5.4.3 30 Calls Traffic Received

The increase of 30 calls in MANET environment with no background flow shown in diagram 50(i), when compared to the other diagrams 50 (ii, iii and iiii) it is clear that there is a significant more number of successful calls that meet the ITU recommendations. It can be seen that 24 calls for approximately 100 seconds meet the ITU recommendation for this length of time and when looked at in relation to diagrams 48 and 49 the trend appears that call performance improves when there is less background flows, in this case no background traffic. When medium and high background traffic flows shown in 50(iii) and 50(iiii) respectively are compared to their corresponding demands under 10 calls (diagram 48 (iii) & (iiii)) and 20 calls (diagram 49 (iii) & (iiii)) scenarios there is a noticeable difference in the number of calls that receive the expected 125 packets. 30 calls despite there being more calls being placed on the network also increases the probability of the number being successful as previously discussed. However under these scenarios it is not the case for example 20 calls with high background traffic demands is able to deliver up to 11 calls for at least 60 seconds meeting the ITU standard, which compared to 30 call scenario was only able to establish 6 calls meeting this standard.

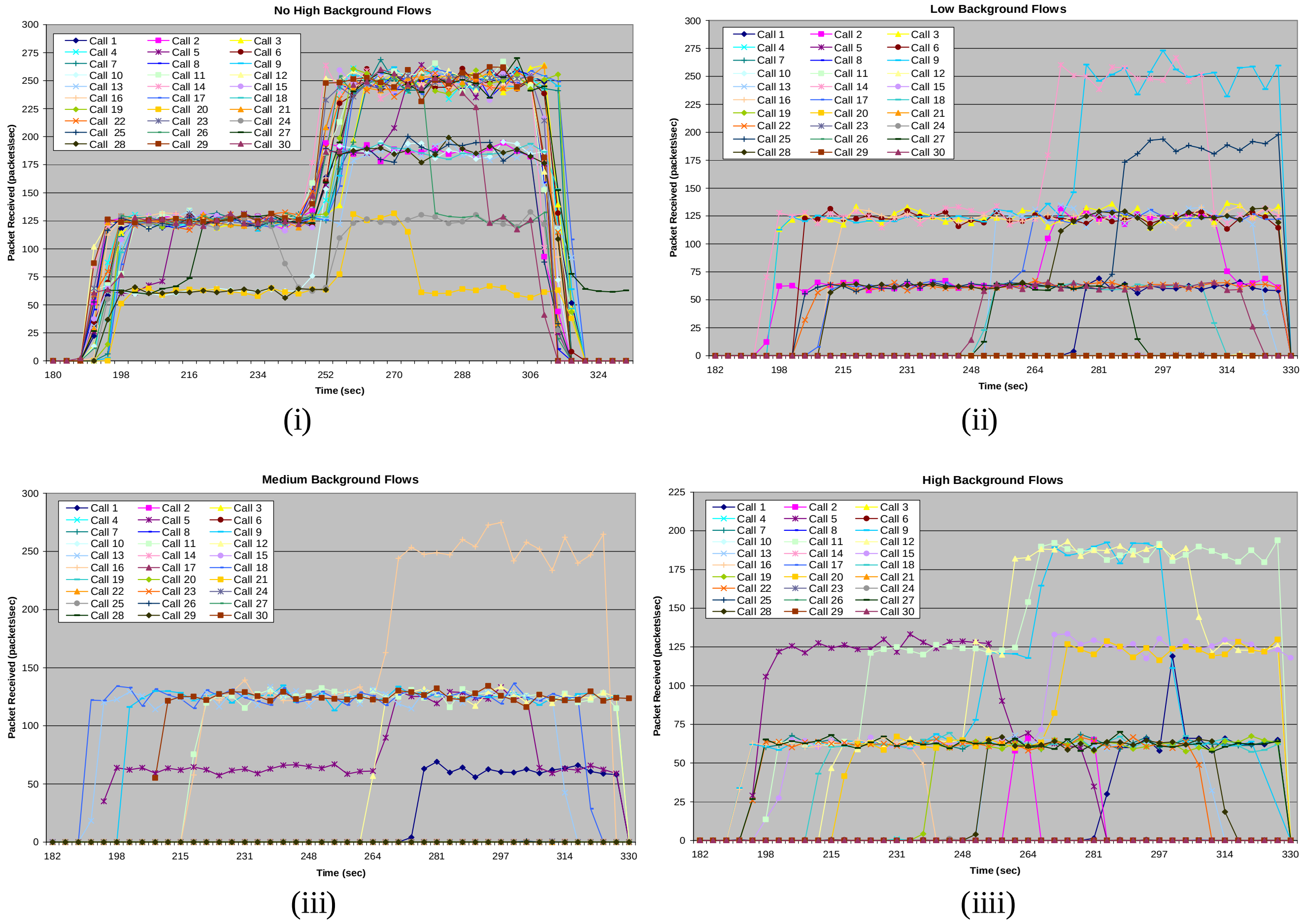
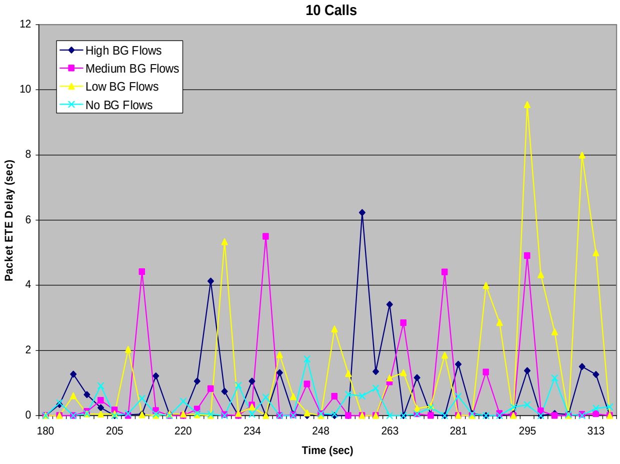


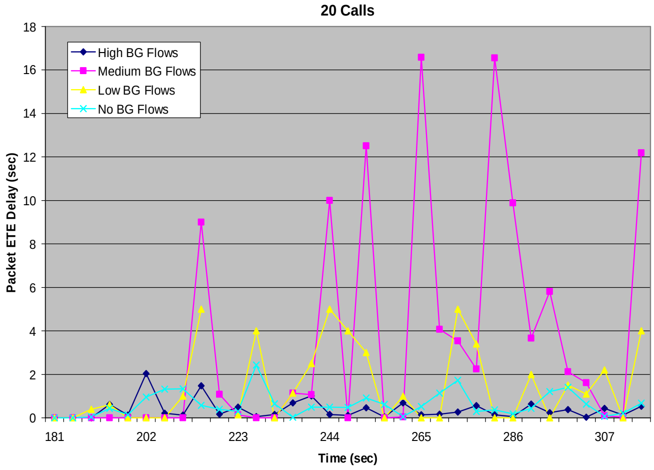
Diagram 50. Received traffic for 30 VoIP calls under: no BG flows (i), 400 low BG flows (ii), 400 medium BG flows (iii) and 400 high BG flows (iiii)

5.4.4 ETE Delay

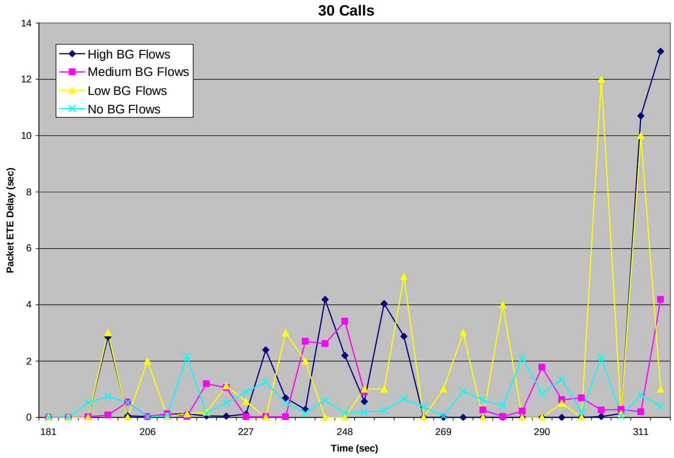
The ETE delay for the average 10, 20 and 30 voice calls under 100 nodes generating different levels of background demands is shown in diagram 51 below. What is noticeable is that despite higher demand background flows producing larger and more frequent packets, thus requiring longer period of time to deliver the same number of packets and more data being routed through the network there is not a clear level of differentiation in regards to voice packets ETE delay times. For example under 20 call scenario, 51(ii) below, it can be seen on average for the duration of the call between 180 and 300 seconds is that the ETE delay between 'No BG Flows' and 'High BG Flows' remains similar with neither going above ~2.2 seconds. They follow a similar pattern of having sharp increases in delay at specific time spots, 230 for 'No BG Flows' and 202 seconds in regards to 'High BG Flows'. This lack of differentiation and that of ETE delay between the implementation of additional calls emphasises the lack of reliability in the network behaviour that can be expected due to MANET's infrastructure less topology. However this also demonstrates a strength of the MANET topology in that despite being put under high demand by both background and calls its dynamic nature and user mobility can ultimately have a positive impact from traffic not being processed by a single device but rather through multiple intermediary nodes that route the packets to the destination. The poorest ETE delay performance is shown by 'Medium BG Flows' when only 20 calls are implemented as shown in 51(ii). It has five clear time periods above 8 seconds at time points: 215, 244, 250, 265 and 286 seconds which when looked at with the AODV route discovery time for the same scenario in diagram 52(ii) below, shows that the experienced high ETE was contributed due to the route to the destinations being lost and having to re-established.



(i)



(ii)



(iii)

Diagram 51. Average packet ETE delay for voice traffic under no BG flows, 100 low BG flows, 100 medium BG flows and 100 high BG flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.4.5 Route Discovery Time

The route discovery time shown in diagram 52 below, illustrates the time for the AODV routing algorithm to discover a path to the destination node for both voice and background traffic flows. This statistic provides information on the dynamic nature of the node movement since periods of high route discovery indicates that nodes are not located in positions that can provide a route to a destination node. Thus also indicating periods of time where the network is not converged since routes to all nodes are achievable. AODV is the implemented routing algorithm and is termed ‘on demand’ discussed in section 2.5. The time to establish a connection between two nodes when compared to a ‘proactive’ routing algorithm would be expected to be higher due to it not keeping a routing table that contains backup routes to destination nodes. Therefore when a like break occurs it is able to re-establish a connection quicker to that of AODV, which requires rebroadcasting ‘RREQ’ packets to its neighbours. As discussed previously the data gathered for MANET is very irregular in the results it produces in regards to performance evaluation. For example when ‘Medium BG Flows’ are compared in 52(i), 52(ii) and 52(iii) it is clear that there is no consistency between performances.

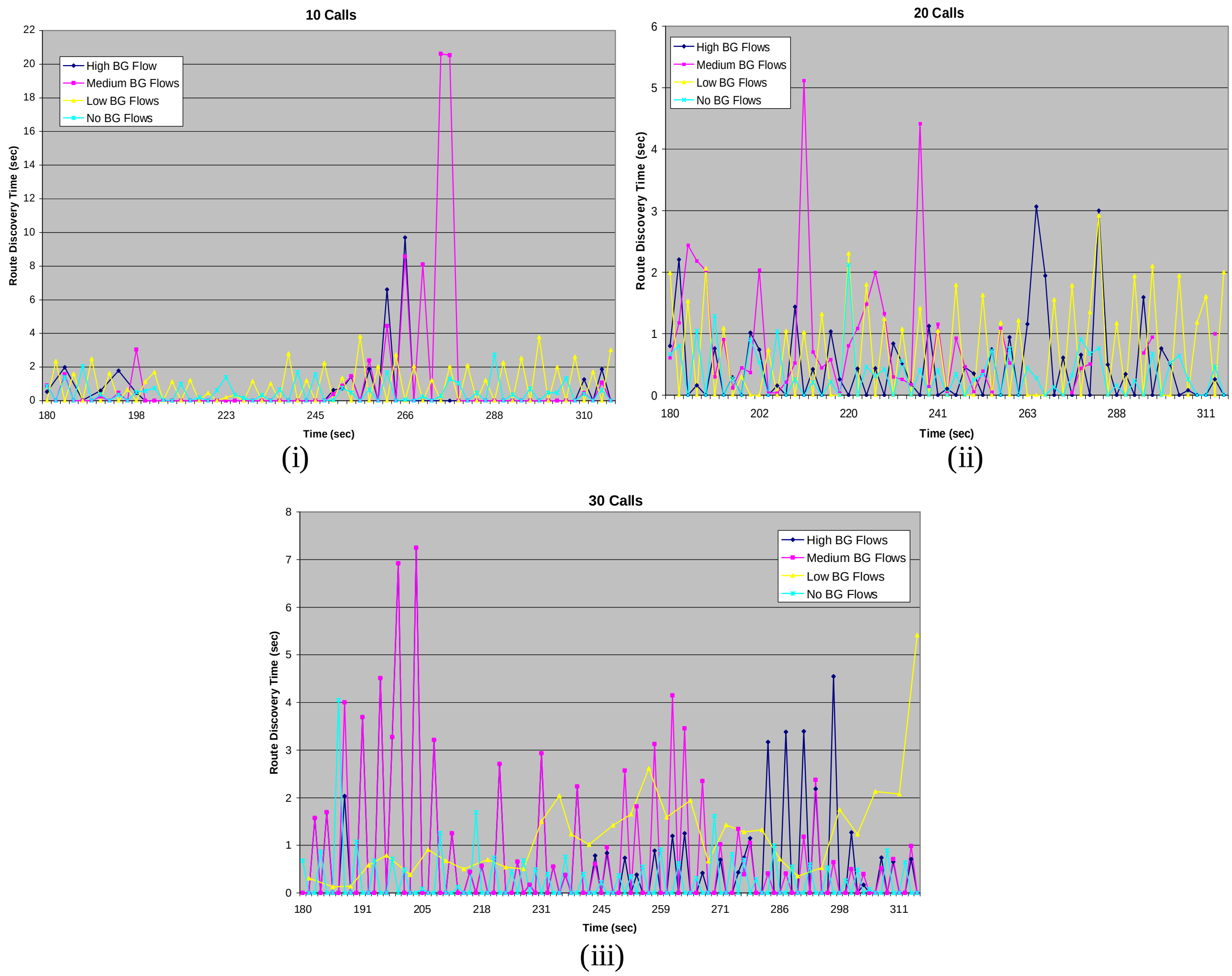


Diagram 52. 'Route Discovery Time' for no BG, 100 low BG flows, 100 medium BG flows and 100 high BG flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.4.6 Conclusion

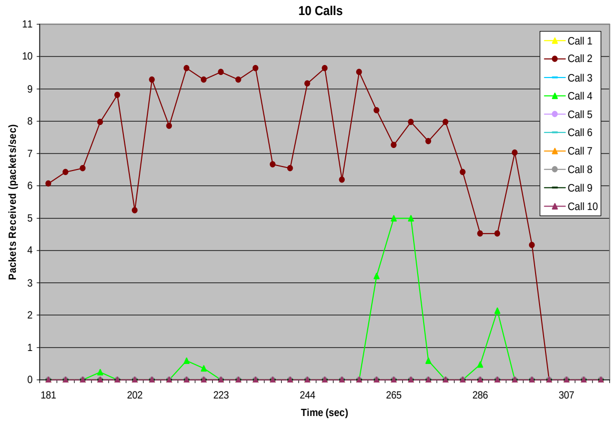
Considerable improvement in the number of calls that meet the ITU recommendation compared to a 200 MANET environment can be seen from diagrams 40-41 to 48-51. This increase can be expressed by the adjustment in the number of nodes within the environment, thus creating a more dense node environment increasing the probability that a route to the destination node being found. When the network has 400 backgrounds flows of any type it does have an effect on the call traffic being received most noticeably when 30 calls with 400 medium and high background traffic flows saw the largest drop in successful connections, compared to 30 calls with no background traffic flows. MANET being an infrastructureless environment and its current lack of guaranteed QoS and route connections is illustrated clearly in diagram 48 (ii), in which within the 400 node environment under low background traffic flows no calls were able to be established for a any length of period and peaking at only 35 packets being received. This is further shown by the ETE delay experienced by calls in diagram 51, which despite the increase in calls there was no noticeable increase in the average ETE delay when the calls were made with no background flows their average delay for the 180 second call duration was below 1 second.

5.5 MANET 200 Nodes: Video Conferencing Calls

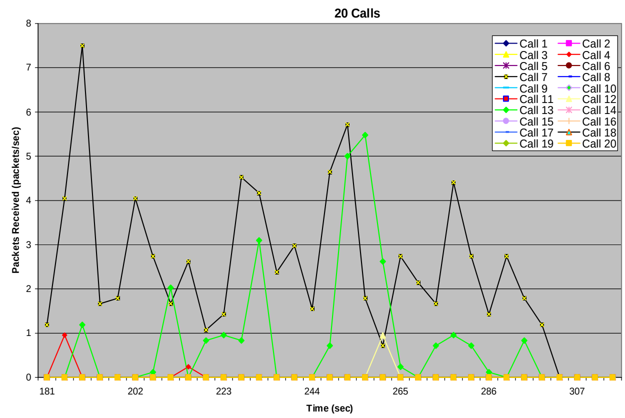
As the results shown in section 5.2 for video conferencing over WAP demonstrated that it was unable to handle a single video conferencing call due to the high packet queue size resulting in buffer overflow. Since MANET environment is dynamic and does not have a single AP which all nodes communicate through but instead routes packets through neighbouring nodes means that the load video conferencing puts on the network will be distrusted across multiple nodes rather than a single device, thus successful delivery of video conferencing calls may be achieved.

5.5.1 Traffic Received with no Background Flows

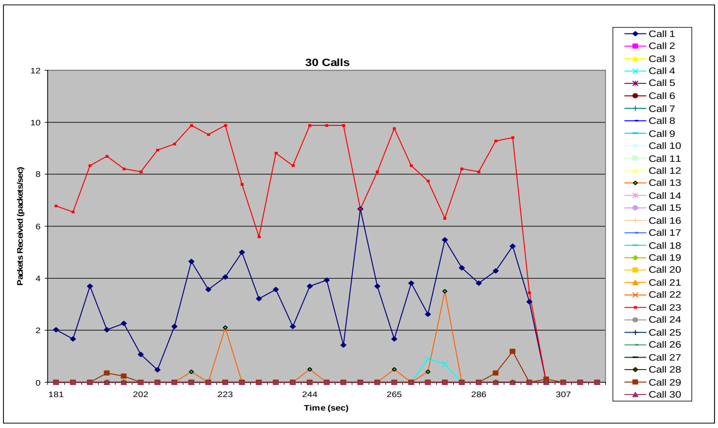
As discussed previously when there is no background flows in the environment it can be seen to be the 'perfect network environment' since there is no other traffic flows except that of the calls. This environment is demonstrated in diagram 53 below, for video conferencing calls received traffic in a 200 node environment for 10, 20 and 30 calls. With the expected packet received being 10 packets, it is noticeable under all three call loads that no calls were successfully received by any of the receiving nodes. Call 2 in 53(i) is the only call within the 10 call scenario that shows significant amount of received packets close to the expected 10 packets a second. However as the diagram shows there is fluctuation throughout the 120 second call period with packets received sharply increasing and decreasing between ranging values of 5 and 9.8 packets a second. Similar call performance level can be seen in the 30 call scenario in 53(iii) by call 23, where for short periods of time the packets received met that expected for a successful call in regards to packets sent and received.



(i)



(ii)

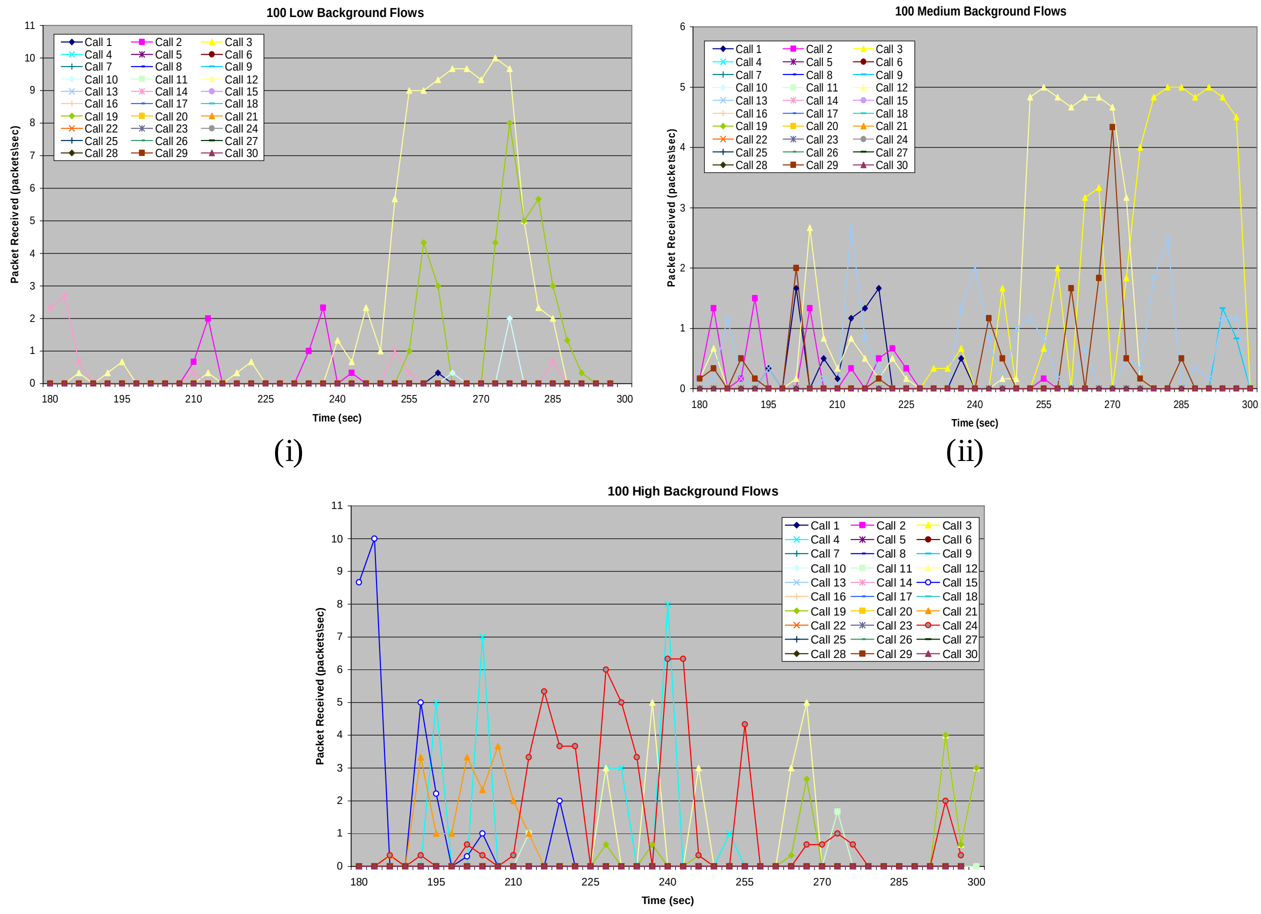


(iii)

Diagram 53. Received traffic under no background flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.5.2 30 Calls Received Traffic with Background Flows

After the review of diagrams in diagram 53, it could be concluded that the results prove that, MANET, like in the WAP environment which nodes use the same buffer size of 256000 bits that dropped packets because of buffer overflow and is the reason for the poor received call rates. However as diagram 54 below shows for 30 calls with 100 different background flows of low, medium and high demands, that in fact there was more video conferencing call packets being received by nodes than that of nodes under no background traffic (54(iii)). This demonstrates the point of MANET distributed topology as performance of an application in this case video conferencing calls, can improve despite there being more traffic being placed on the network, which in this scenario is background traffic. This distributed topology as a result means that the performance and as documented in the performance of VoIP in a MANET environment above is significantly dependent on node positioning for a route be established and maintained between source and destination. This can lead to a scenario where a single node is in a position to provide a route to multiple destination, which in this case a bandwidth intensive application such as video conferencing and the node buffer is only able to store approximately 2 seconds worth of this type of traffic. Thus it can be seen that calls that received high number of packets that of 5 and above either had temporary direct connections with the sending node or the intermediate hop nodes along the route did not have other video conferencing calls being sent across it.



(i) (ii) (iii)

Diagram 54. Received traffic for 30 calls for 100: low BG flows (i), medium BG flows (ii) and high BG flows (iii)

5.5.3 10 and 20 Calls Received Traffic with Background Flows

Diagram 55 below shows the average voice traffic received by receiving nodes for the duration of the 120 second calls under different background demands: low medium and high by 100 nodes within the environment. It is noticeable when both diagrams 55(i) and 55(ii) are compared that under the medium background demand is that they are the only calls that for the majority of the call period receive traffic, despite only being a maxim of 1.6 to 1.7 packets for a duration of 5 seconds at the time period 285 seconds as shown in diagram 55(ii). The irregular pattern of receiving packets under the different demands further demonstrates as already well documented in this report the lack of reliability that MANET topology can provide. Under the six scenarios shown in the diagram below on average receive less than a single packet per second, which could be the result of a route to destination being unable to be established or there is a bottleneck in the network where a node is intermediate hop for large number of routes and thus dropping packets due to buffer overflow.

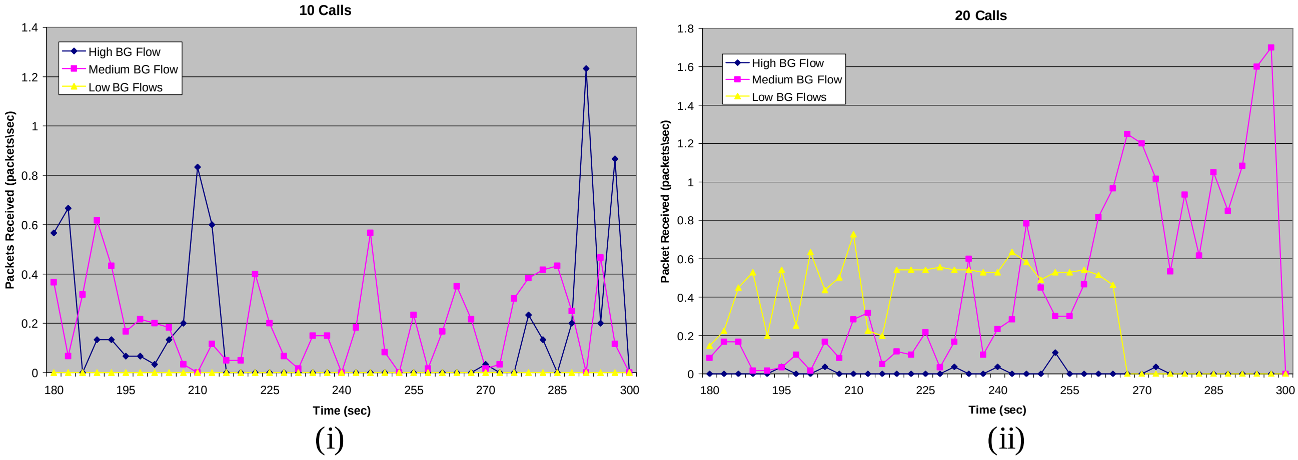


Diagram 55. Average received traffic for 100 low, medium and high background flows for: 10 calls (i) and 20 calls (ii)

5.5.4 ETE Delay

With the ETE delay as shown in diagram 56 below, containing values of 0 for large periods of time for calls 10, 20 and 30 under varying 200 background flow demands is the result of the lack of packets being received by nodes in regards to received video traffic. Since no packets are being received by nodes thus there is no ETE delay being produced and therefore being recorded by a value of zero. In each diagram there is noticeable spikes in the ETE delay that range from 2 seconds up to 5 seconds. An example of this is in 56(ii) at the time point 195 seconds where there is a sharp increase in ETE delay from approximately 0.05 to 2.5 seconds and at the next data sample spiking back down to approximately 0.05 seconds. The sharp increase can potentially be explained by routes between source and destination being broken so a new route will have to be established but also because packets will have to wait in intermediate hop nodes buffers before being sent and therefore adds to the delay experienced.

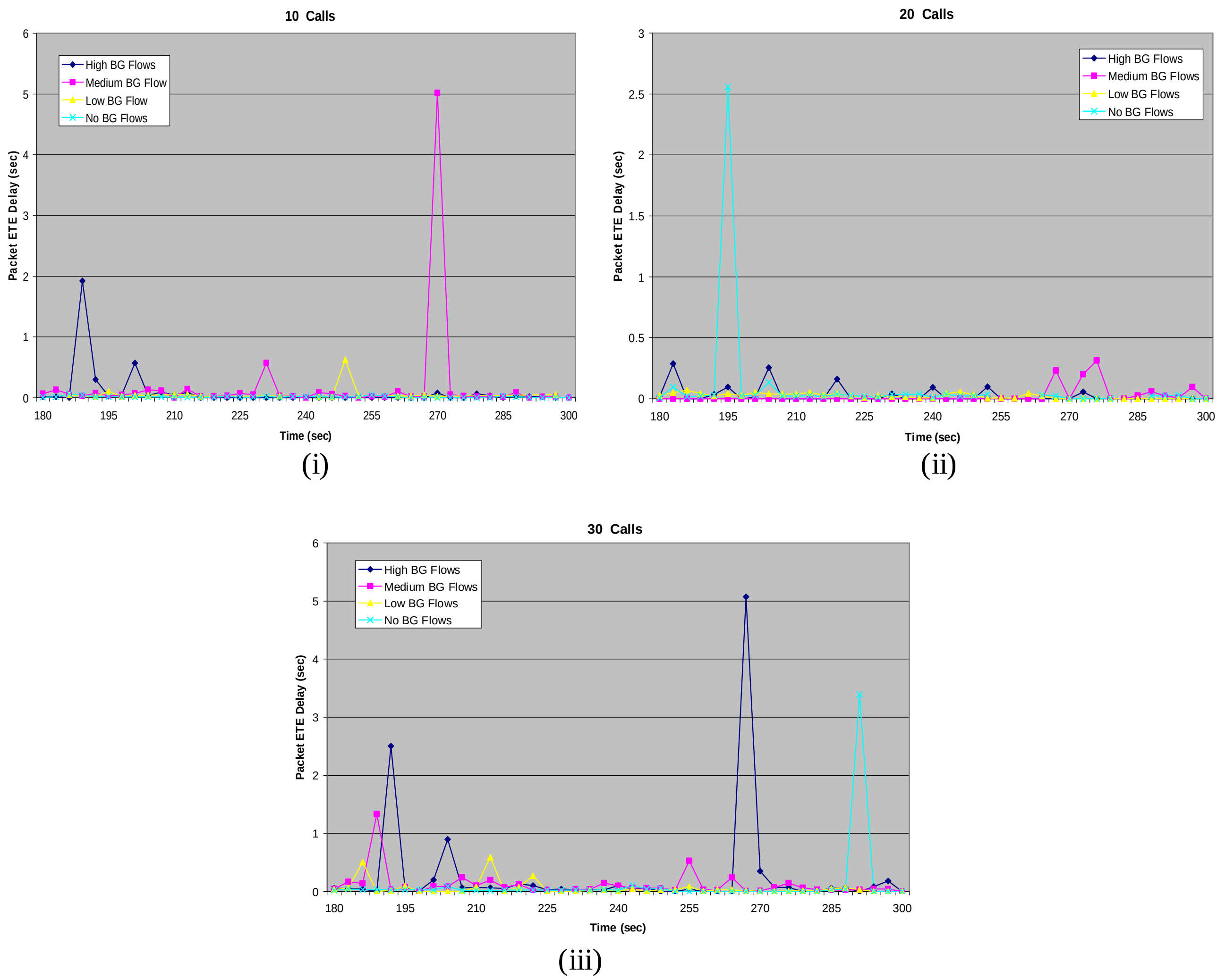
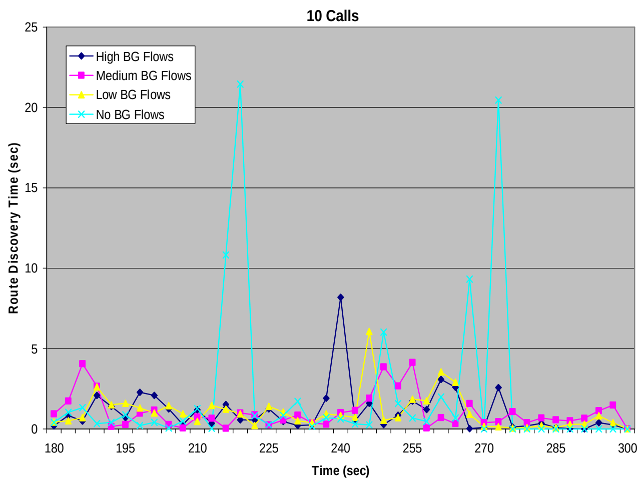


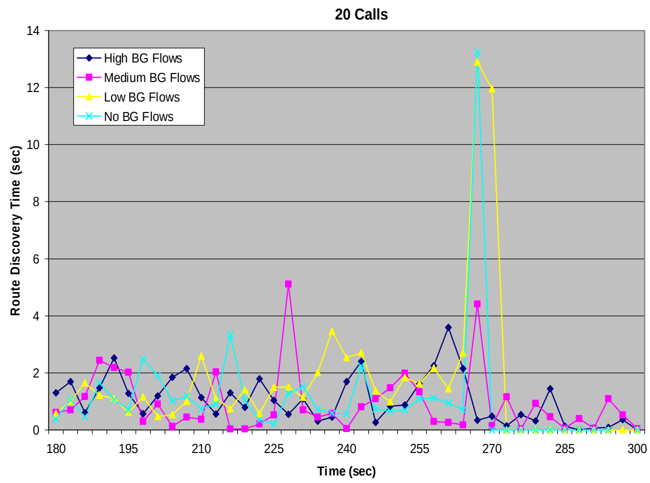
Diagram 56. Average packet ETE delay for video traffic under no BG flows, 200 low BG flows, 200 medium BG flows and 200 high BG background flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.5.5 Route Discovery Time

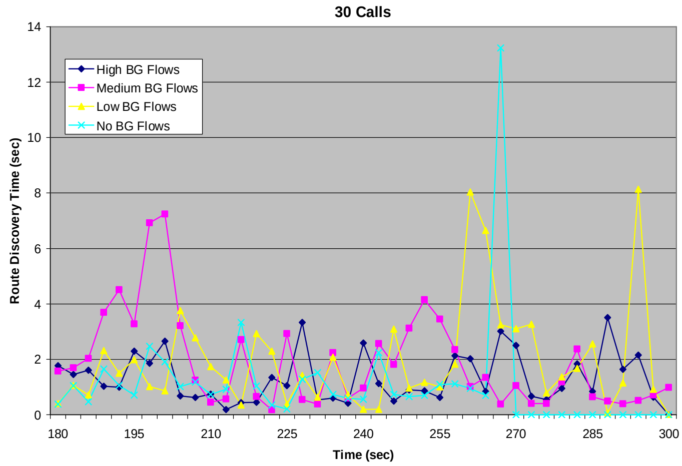
The route discovery time for 10, 20 and 30 calls under different types of 200 background flow demands during the time period of 180 and 300 seconds is shown in diagram 57 below. Since the route discovery time includes the time for both voice and background flow demands it provides an overview of the performance of the network and route path changes as a result of node movement. As discussed previously since AODV uses its own 'RREQ' packets to first establish a connection before sending user data and routing packets being tagged with the highest QoS priority means that the discovery time should not be effected by the traffic demand type. This is indicated in all three of diagrams 57, with 'no BG flows' peaking at the highest route discovery time for example at time period of 215 to 225 seconds in 57(i) when there is a sharp spike from 0 (thus no routes having to be established) to 11 at the next time sample increasing to 22 seconds at the following sample and then steep decrease in discovery time 1 second. These changes in discovery times demonstrates not only the dynamic nature of MANET but also when a environment has a low node density then specific node movements can result in these high discovery times occurring. This is significant in the case when a node is positioned that it is the intermediate hop node for multiple different connections and therefore could be the bridging node between two formed groups of nodes. Therefore this node's movement can result in the bridge failing and all the connections between groups being dropped and a new route established resulting in route discovery time.



(i)



(ii)



(iii)

Diagram 57. 'Route Discovery Time' for 200 high, medium, low and no background traffic flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.5.6 Conclusion

As was identified in the WAP environment (section 5.2) a single video conferencing call was unable to be made in a 200 node MANET environment with no background flows. The test data showed several calls that received half the expected packets rate of 10 second. This was most noticeable in the scenario in diagram 50(iii) which had 30 calls with 100 nodes producing high background traffic flows, which when compared to 30 calls with no background flows there was an increase of 3 calls receiving 3 or more packets. This comparison demonstrates the dynamic nature of MANET and the traffic flow distribution through the topology. By routes to destination being established on an 'on demand basis' means that the routes to destination nodes can change as nodes move within the environment. Thus in certain instances single nodes may be in a position within the environment, which they will be routing packets for several data and call flows. As a result as documented the nodes are only able to buffer 2 seconds of video traffic and therefore packets will be dropped due to buffer overflow. As with VoIP over a 200 node dense MANET environment the number of calls that receive zero packets may have been the result of being in a position within the environment during the 180 and 300 seconds call period that they had no neighbouring nodes. Thus why testing video conferencing in an environment containing 400 nodes was required and the results are discussed in the following section.

5.6 MANET 400 Nodes: Video Conferencing Calls

As earlier discussed to increase the probability of nodes obtaining the optimum number of neighbouring nodes that research by Royer et al (2001) concluded that seven to eight neighbours was the optimum number, the additional 200 mobile nodes potentially will aid in proving this. Thus there could be a noticeable difference shown by the video conference calls meeting the ITU calls recommendations than that of the MANET 200 node scenario.

5.6.1 10 Calls Traffic Received

Diagram 58 below, shows the video packets received by 10 calls under 200 background flows in which the flow demands characteristics are changed to include low, medium and high flow demands. Through changing these demand characteristics it can be seen from the diagrams in 58 is that it does not have an effect on the traffic received. This can be stated because despite no calls meeting the expected 10 packets per second in any of the scenarios when 58(i) and 58(iiii) are compared there is not a noticeable difference. Thus shown by 58(iiii) where under half the node population sending large background traffic flows there was more call connections established and maintained for the duration of the 120 second call period between 180 and 300 seconds. What can be noted from diagram 58(ii) is that ‘call 2’ during the first 40 seconds manages to achieve and maintain receiving of between 2 to 5 packets a second until 230 second time point, which its call then drops to 0 packets a second for the vast majority of the rest of the call time period. This can be seen to potentially be due to the source, destination or route hops nodes having moved into a position in which a route to the destination can no longer be establish or a nodes buffer is over flowing.

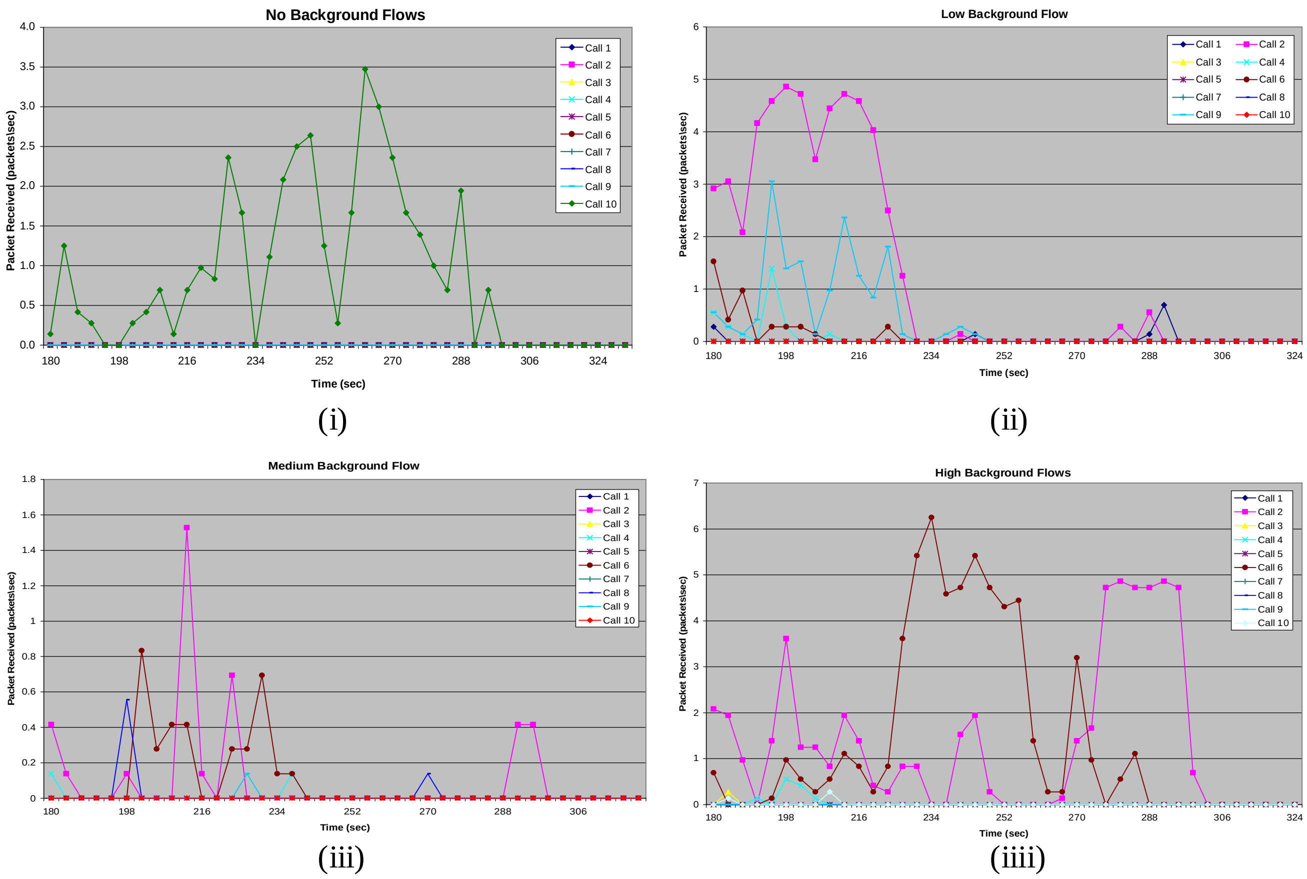


Diagram 58. Received traffic for 10 video conferencing calls under: no BG flows (i), 200 low BG flows (ii), 200 medium flows (iii) and 200 high BG flows (iiii)

5.6.2 20 Calls Traffic Received

The deployment of 20 video conferencing calls placed on a network that has half of the environment node population generating 3 different background flow demands (low, medium and high) the results of which are shown in diagram 59 below in regards to video packets received by the called nodes. It is noticeable as when 10 calls are deployed, discussed directly above, is that the introduction and changing of the demands of background traffic flows has no visible effect on calls from the information shown in diagram 59. Where under no background traffic flows only call numbers 11 and 16 peak >=1.5 packets per second (59(i)), when in the case of 200 low background flows(diagram 45 (ii) call numbers 4,15 and 18 achieve this. Further to this point ‘call 10’ in 59(iiii) shows a constant receiving rate of 5 packets for a duration of 30 seconds starting at 198 seconds. This illustrates and could be concluded to be due to it staying at a constant value of 5 was the result of a node or nodes along the path being unable to handle the number of packets it was receiving due to video conferencing high packet demands or traffic being routed from other traffic demands. This can be further proven by the node buffer limited to only buffering 2 seconds of a video conferencing call.

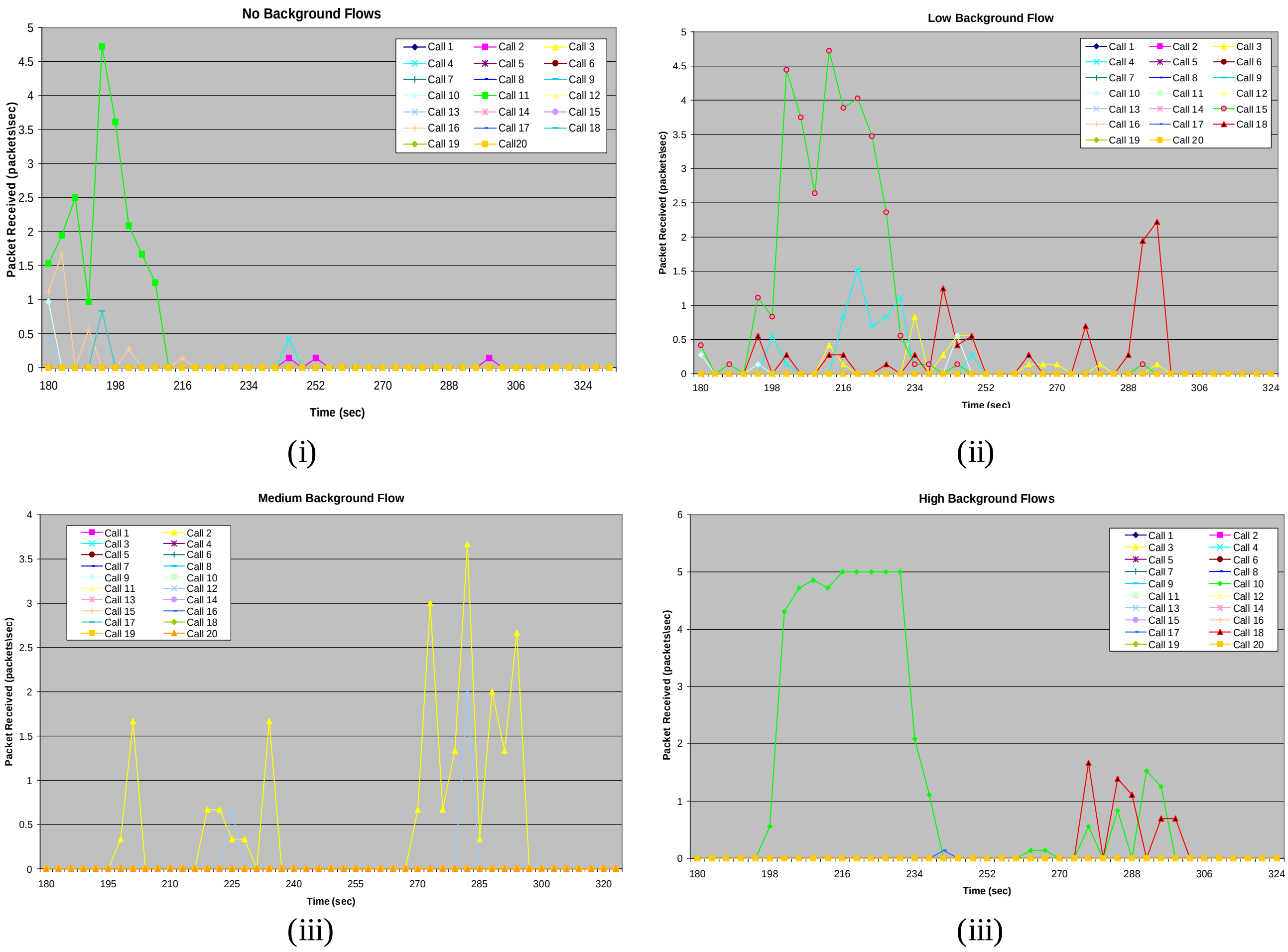


Diagram 59. Received traffic for 20 video conferencing calls under: no BG flows (i), 200 low BG flows (ii), 200 Medium flows (iii) and 200 High BG flows (iiii)

5.6.3 30 Calls Traffic Received

The dynamic nature of a MANET environment, which has been demonstrated and discussed above, has shown that by its nature means its behaviour in regards to producing results that would be expected from information gathered from previous scenarios can not be assumed. This point is demonstrated by diagram 60, where 30 calls being placed under the same background conditions as in calls 10 and 20 (section 5.6.1 and 5.6.2 respectively) however it does not follow the trend that was seen of networks with background flows. The result was that more calls received higher packets per second for longer periods of time compared to that of the calls under no background flows. Thus in the case of 30 calls being placed across the network it can be seen from diagram 60 that under no background flows (diagram 60(i)) achieved a steady packet rate of 4.5 packets per second for the duration of 40 seconds during the time period of 260 and 300 seconds. When compared to diagram 60(ii), (iii) and (iiii) there are no other calls that achieve this level of packet delivery with only diagram 60(iii) peaking at 225 seconds achieving a received packet rate of 4, lasting a duration of approximately a single second before decreasing sharply at 235 seconds where it reaches a lower packet rate of only 1.4 packets.

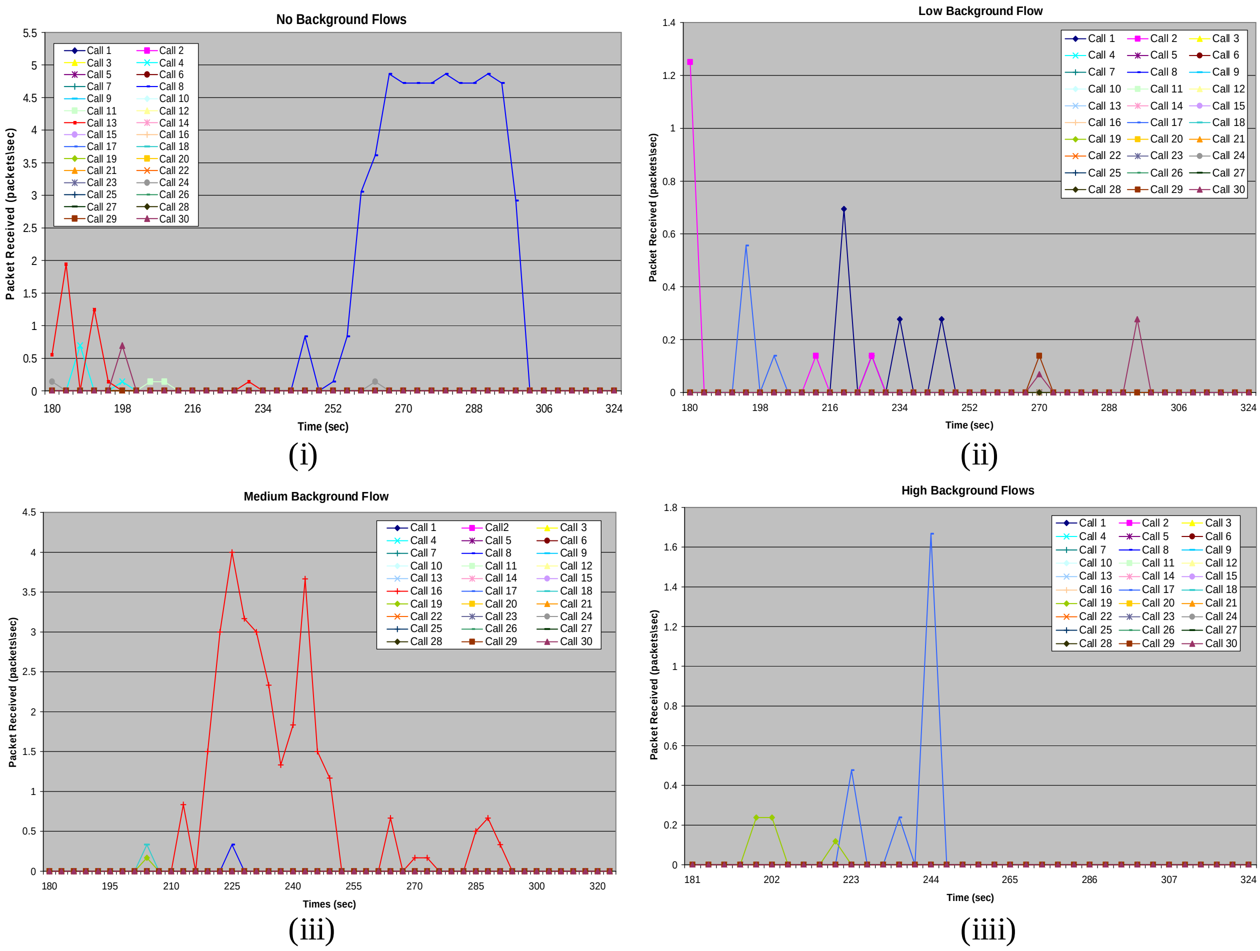
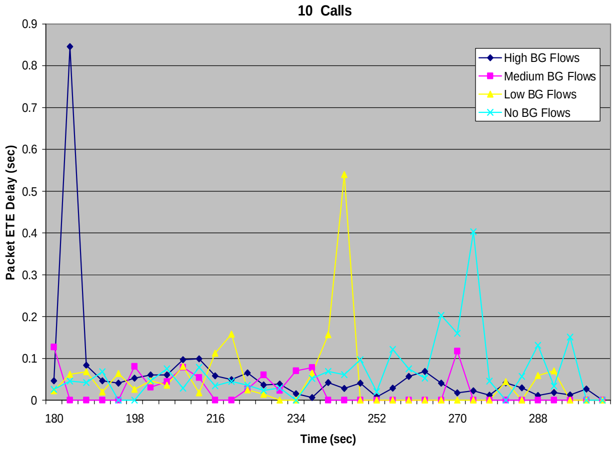


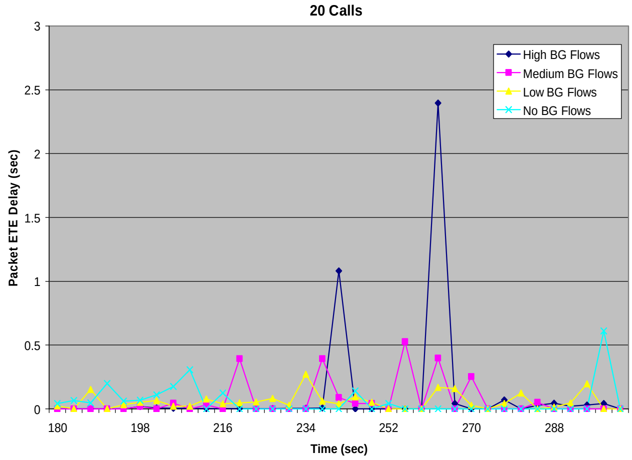
Diagram 60. Traffic received for 30 video conferencing calls under: no BG flows (i), 200 low BG flows (ii), 200 Medium flows (iii) and 200 High BG flows (iiii)

5.6.4 ETE Delay

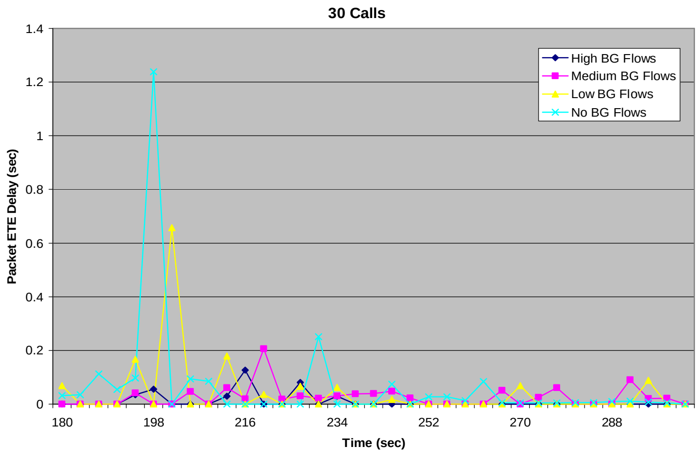
The ETE delay that diagrams 61 shows is for 10, 20 and 30 video conferencing calls that are placed in the environment that contains varying types of 200 background flow demands. In the three diagrams it is noticeable that despite when calls are made under no background demands in the environment they still experience steep increases in ETE delay as shown in 61(iii) at time point of 198 seconds where there is a sharp increase from 0.1 to 1.2 seconds in delay. These increases are also shown by low and high background traffic flows in diagram 61(i) at time points 240 seconds and 185 seconds respectively. Despite traffic received under each scenario shown in diagrams 58-60 above being below the expected 10 packets the ETE delay of these packets is for a large majority of scenarios below the recommended 150 millisecond. For example the 10 call scenario, diagram 61(i), shows that it was only when there was spikes in the ETE delay that the delay went above the recommended 150 milliseconds. When these spikes occur, the user would experience prolonged periods of silence thus it may cause the user at the other end of the call to speak at the same time since they hear silence.



(i)



(ii)



(iii)

Diagram 61. Average packet ETE delay for voice calls under no BG flows, 200 low, medium and high BG flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.6.5 Route Discovery

Diagram 62 below demonstrates the route discovery time of 10, 20 and 30 calls placed over the 400 MANET environment which has 200 background flows of varying demands. When the below diagram is compared to that of all the other MANET environments both 200 and 400 node population and VoIP and video conferencing calls their average discovery time except for the period of time where there is high spikes can be said to be on average between 1 to 3 seconds. This is of interest as it demonstrates how node population does not appear to have a noticeable difference when additional 200 mobile nodes are implemented into the environment. This can be argued since there are more nodes as in the case of the 400 node environment there is an increased probability a route will be discovered quicker but can also result in more hops having to be routed across to find the destination. When compared to 200 nodes the same level of route discovery time can be argued as there is less hops to route through to reach the destination node but is offset by the decrease in probability of nodes being in positions to provide complete network coverage across the whole network environment. Three flows that are of particular interest is the time period in diagram 62(i) which during the time period of 252 and 270 seconds, background traffic flows high, medium and low all follow the same performance increase and decrease pattern during this time period. Though the performance of the three traffic flows are different leading up to this time period it could be viewed and as further discussed in section 6.7.2 that the mobility of the nodes followed the same movement patterns. This however cannot be fully declared as the cause since all the simulation scenarios were carried out the same number of times and it is only in the scenario in which there are three flows performed so similar during a specific time period.

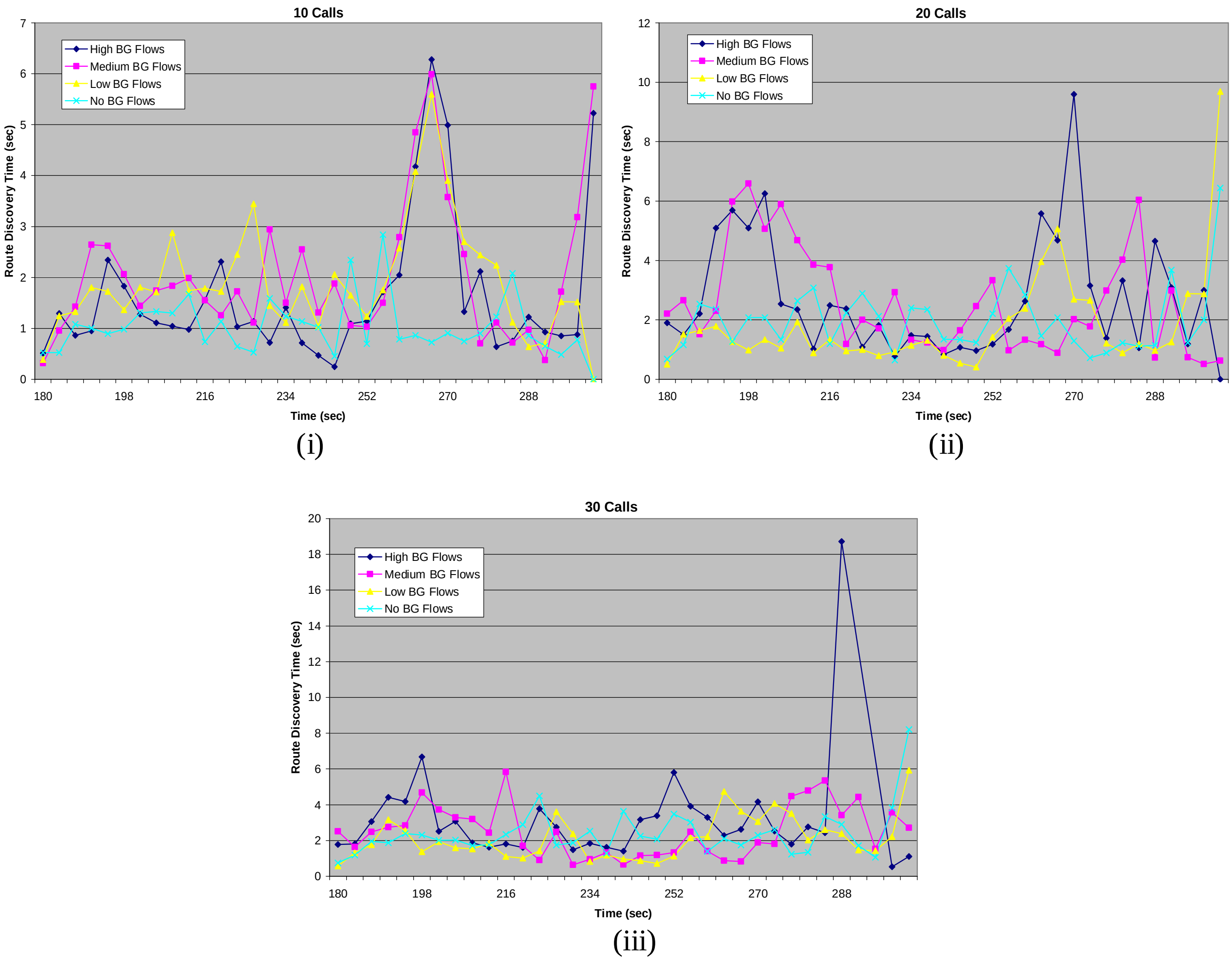


Diagram 62. 'Route Discovery Time' for no BG flows, 200 low, medium and high BG flows for: 10 calls (i), 20 calls (ii) and 30 calls (iii)

5.6.6 Conclusion

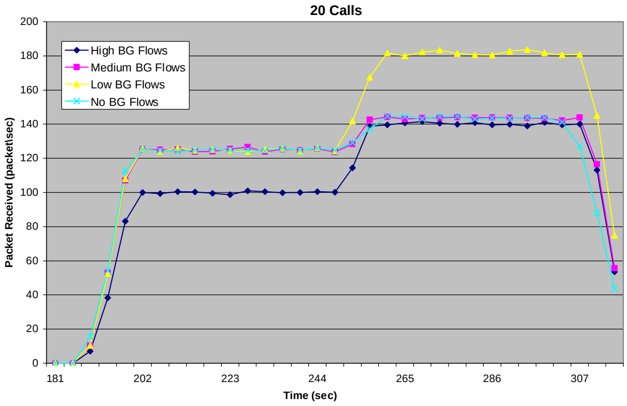
What can be concluded from the presented results in regards to video conferencing over a MANET topology consisting of 400 nodes in regards to video conferencing packets received is that there is no noticeable difference in improvement from having additional 200 mobile nodes within the environments. It may have been expected through the additional nodes, that it would increase the likelihood of a node having neighbouring nodes and routes to destination nodes but from the results this is not the case. This poor performance can be seen to reflect the dynamic nature of the topology through there being no distinguishable difference between the introduction of increase of calls and background traffic demands but essentially it is the video conferencing application itself. As documented in section 5.1 under the WAP topology a single video conference call was not possible through a dedicated routing device under a single call scenario with no background traffic flows. Thus it could be expected and is shown from diagram 60(i) that improvement on the WAP scenario performance was achieved. The introduction of additional nodes does not have a noticeable impact on the route discovery time possibly for a number of reasons as discussed in this section.

5.7 ESS Small: VoIP Calls

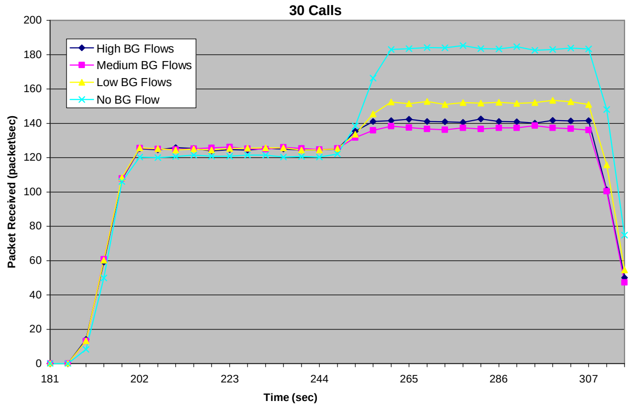
As discussed in section 2.1.3, ESS expands on a single WAP environment through the deployment of multiple WAPs positioned with overlapping signal areas with a common route protocol to update routing tables, which in this project is OSPF. Below are the results of the smaller of the two ESS environments with the deployment of 80 nodes and 12 WAPs with VoIP being the call type.

5.7.1 Traffic Received

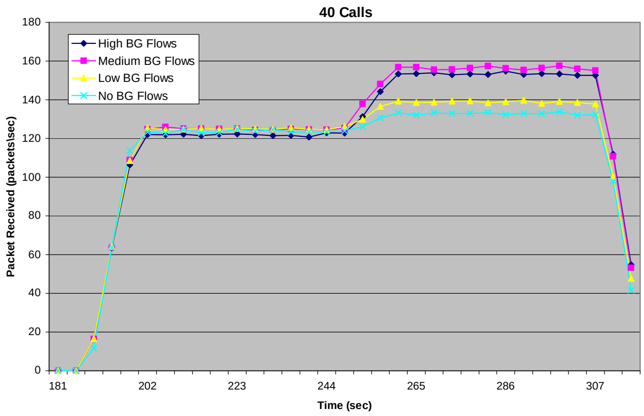
The average call packets received per second by nodes when 20, 30 and 40 calls are placed across the ESS network under 40 nodes (half the node population) with varying levels of background flow demands is show in diagram 63 below. There are three main things that are noticeable from the results, the first point is shown in 63(i), when 20 calls are placed under heavy background flows that during the time period of 202 and 250 seconds the traffic received is below the recommended 1% drop that of 125 packets per second expected. When this heavy background flows is compared to that under 30 and 40 calls it shows there is a noticeable difference by the fact they do not drop packets during this period of time. This however could be due to node mobility and positioning within the environment with a particular WAP having multiple nodes associated with it. However this could be the result of point two, which is the result of the increase in traffic received experienced in all scenarios at 250 second time spot in diagram 63. This increase also occurred in section 5.3.2 and will be discussed below in section 6.6. The third point is that from high background flows placed in a 20 VoIP call scenario, all other calls during the time period of 202 and 250 seconds met the ITU 1 % drop. This can be seen to be achievable since the network load is highly likely to be spread over multiple routers since each router provides coverage to specific areas of the wireless network. Thus as the nodes moves through the environment it will undertake the 'handover' process where it associates itself with a new WAP due to receiving a stronger signal



(i)



(ii)

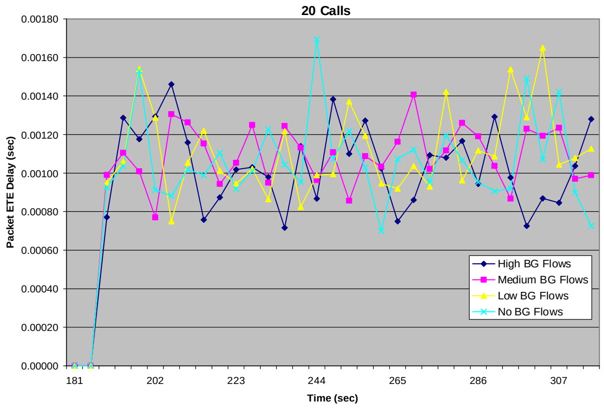


(iii)

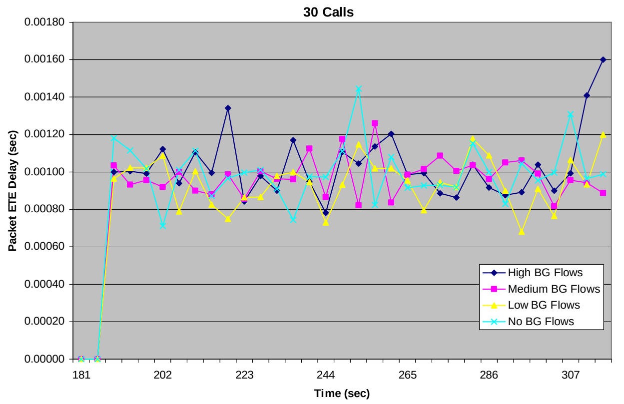
Diagram 63. Average received traffic under no BG flows, 40 low, medium and high BG flows for: 20 calls (i), 30 calls (ii) and 40 calls (iii)

5.7.2 ETE Delay

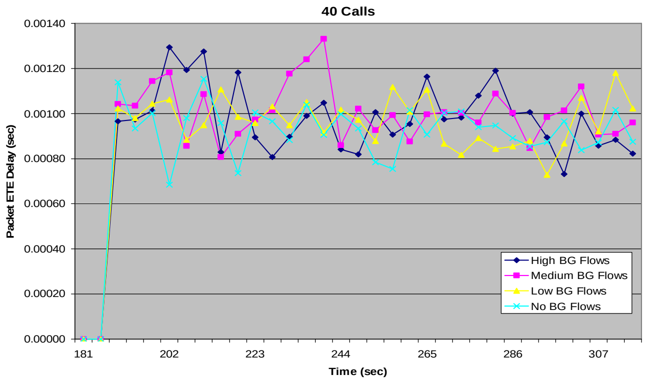
Another metric of the ITU recommendation is the ETE delay of calls which is shown in diagram 64 below for 20, 30 and 40 calls under varying 40 background flow demands. The three sub diagrams of diagram64 shows the average delay that crosses through 0.001 seconds which is significantly lower than that of the ITU recommendations. This shows that the QoS mechanism that is in place is performing as it was designed for by prioritising voice traffic over background traffic flows even with various packet sizes and packet rates. From diagrams 64(i) and 64(ii) it is notable that both the no background flows and low backgrounds flow performance is the best with the lowest average ETE delay for the duration of the 120 second. However the same background loads under 20 VoIP call scenario, diagram 64(ii) sees their performance less consistent with variations in delay ranging in the case of no background flows from 0.00165 to 0.0070 seconds. This fluctuation could be seen due to user movement and the number of nodes per WAP as discussed in the above section in regards to received packets.



(i)



(ii)

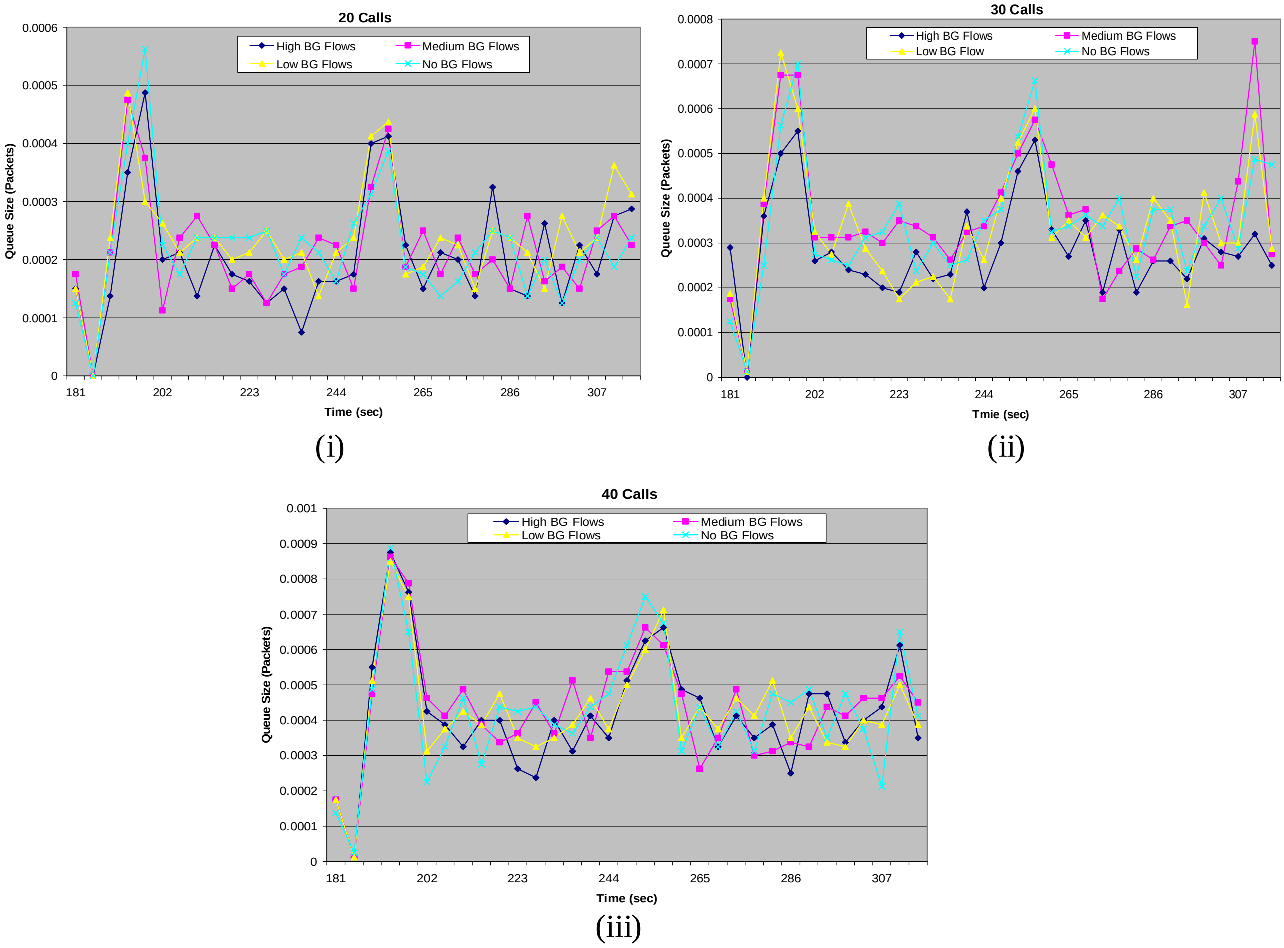


(iii)

Diagram 64. Average packet ETE delay under no BG flows, 40 low, medium and high BG flows for: 20 calls (i), 30 calls (ii) and 40 calls (iii)

5.7.3 Voice Queue Size

The queue size for voice is an important statistic as demonstrated in VoIP over WAP in section 5.1 which highlighted how the implementation of a tenth call significantly increased the voice queue size. Diagram 65 below shows this statistic in regards to the average queue size of the 6 routers for VoIP calls 20, 30 and 40 placed over the small ESS topology in which there is 40 varied sized background traffic flows. What is noticeable is as the number of calls increases from 20 to 30 the average queue size during time periods 202 to 244 seconds and 265 to 300 seconds increase by 0.0001 and this increase is the same between 30 and 40 calls that increases from 0.0003 to 0.0004. These small queue sizes shows that VoIP in regards to 40 calls in a 6 WAP EES environment does not cause any buffer overflows due to the load being spread over multiple routers further to this is VoIP G.711codec delivers small sized packets but at a high rate. This is due to humans being sensitive to delay when communicating in a conversation; therefore it is not practical to store voice data into large packets.



(i) (ii) (iii)

Diagram 65. Average voice queue size under no BG flows, 40 low, medium and high BG flows for: 20 calls (i), 30 calls (ii) and 40 calls (iii)

5.7.4 Conclusion

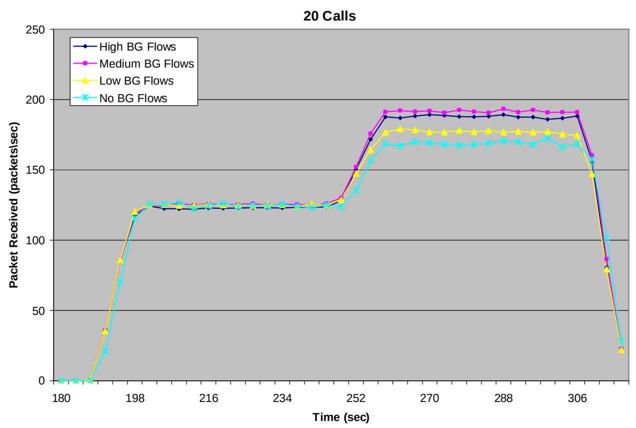
For the duration of VoIP calls during the time periods 202 to 250 seconds demonstrates that 40 calls can be achieved under this ESS topology in regard to VoIP calls with different background traffic flow demands in the background. The results from time period from 250 seconds to the end of the call are discussed in section 6.6 below. With the ITU recommendation of 150 milliseconds in regards to ETE delay diagram 64 clearly shows that this can be met with the highest ETE delay peaking from all scenarios when 20 calls were placed with no background flows reaching 0.0016 seconds. The nature of ESS and user mobility distributing the load of the calls over multiple WAPs and the VoIP codec combined results in the WAP voice queue sizes being so low and reduces packet delay and probability of dropped packets.

5.8 ESS Large: VoIP Calls

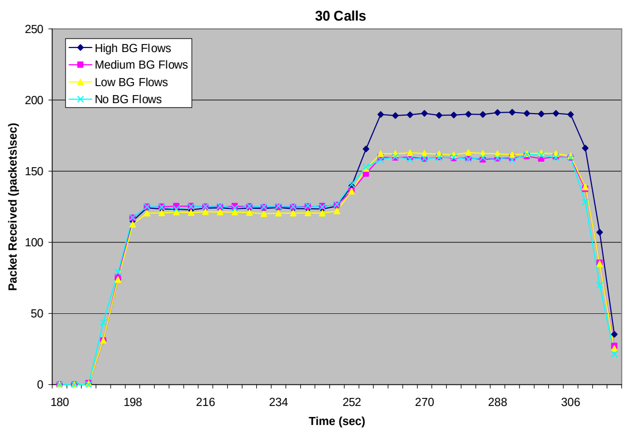
The data presented in the following sections below, presents the results for VoIP calls placed over the large ESS topology that contains 22 WAPs.

5.8.1 Traffic Received

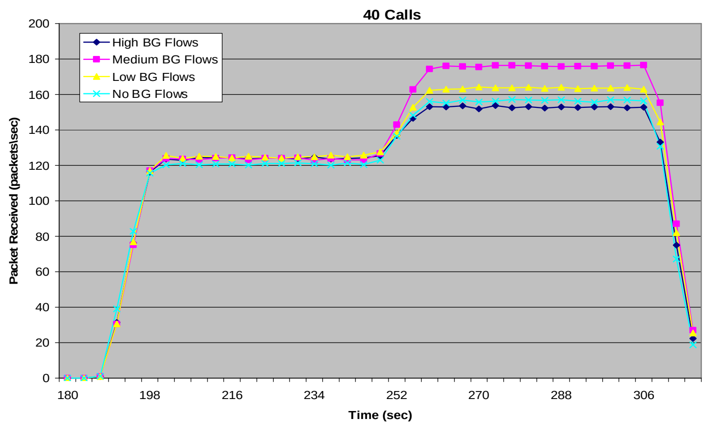
The average packets received by the receiving called nodes is shown for the duration of the 120 seconds, which starts at 180 seconds and ends at 300 seconds shown in diagram 66. The results show that once the calls have been established at 190 seconds that for the next 60 seconds all 10, 20 and 30 calls scenarios meet the ITU recommendation of 1% drop since they receive the expected 125 packets a second. During this period of time there is no clear differentiation of the change in packets received when placed under different background flow demands. This demonstrates as discussed in the small ESS environment the distribution VoIP load between multiple routers and the implementation of QoS mechanism thus the traffic is not trying to be sent through a single device as in the case of WAP. At the 252 seconds mark again all scenarios experience an increase in packets received to the end of the calls period and the reason for this is discussed in section 6.6.



(i)



(ii)

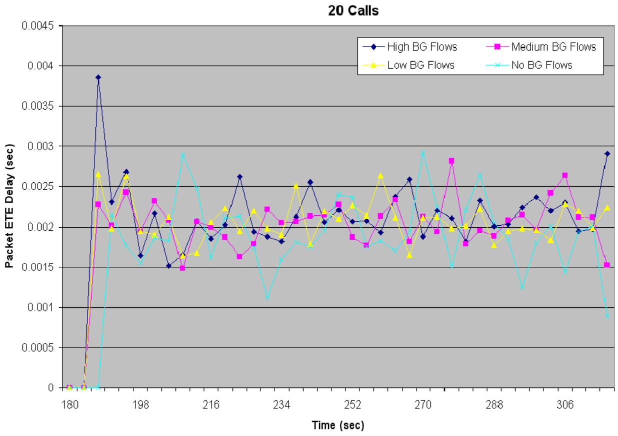


(iii)

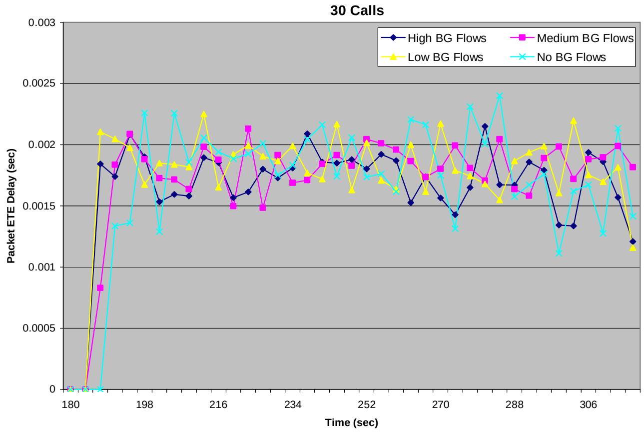
Diagram 66. Average received traffic under no BG flows, 40 low, medium and high BG flows for: 20 calls (i), 30 calls (ii) and 40 calls (iii)

5.8.2 ETE Delay

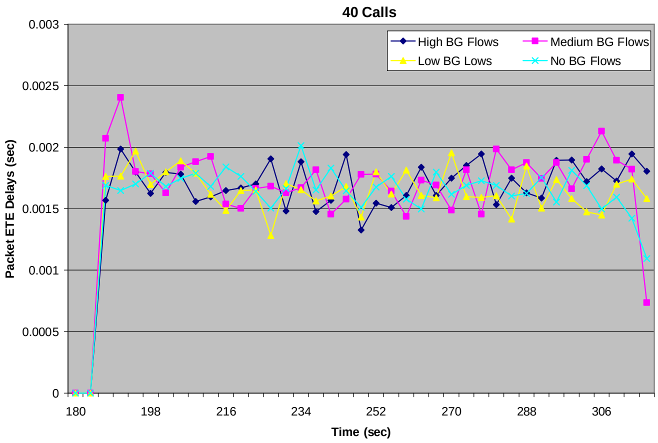
The ETE delay is another metric that the ITU recommends to use to classify if a call would be acceptable to a person. Diagram 67 below, shows the ETE delay for 20, 30 and 40 calls over 40 background flow demands that vary in packet size and packet rate. The results show that all calls meet the ITU recommendation of the 150 milliseconds under all background flow demands. Diagram 67 shows that 30 and 40 calls average ETE delay is between 0.0015 seconds and below 0.002 seconds however the lowest number of calls, 20 calls, is between 0.0018 seconds and 0.0024 seconds. Despite the scenario containing less calls but having a higher average ETE delay shows the impact that mobility has in a large environment. Thus a higher number of calls could have been routed across the network that required a high number of hops to reach the destination node and therefore would result in this higher ETE delay being experienced.



(i)



(ii)



(iii)

Diagram 67. Average packet ETE delay under no BG flows, 40 low, medium and high BG flows for: 20 calls (i), 30 calls (ii) and 40 calls (iii)

5.8.3 Average Voice Queue Size

The average queue size for the duration of the 120 second phone call for voice traffic under 40 different types of background flows are shown in diagram 68 below. As would be expected the type of background flow has no impact on the queue size of the voice due to, as previously discussed, background traffic being stored in a separate queue. Similar to the results of ESS small voice queue size in diagram 65 above, as more calls are made across the network the average queue size increases. There is on average an increase of 0.00002 packets a second as shown in diagram 68(i) on average being 0.00002 to 0.00004 packets a second, 68(ii) 0.00004 to 0.00006 a second and

68(iii) 0.00005 to 0.00007 packets a second. The increases in queue size at the specific point of 260 seconds indicate the time point where the received packets by nodes increases as shown by the data in diagram 66 above. The drop in queue size after this point is not a result of packets being dropped due to the test data identifying that there was no bits dropped due to buffer overflow. Thus it can be seen to be due to the packets being received levelling off to a constant rate.

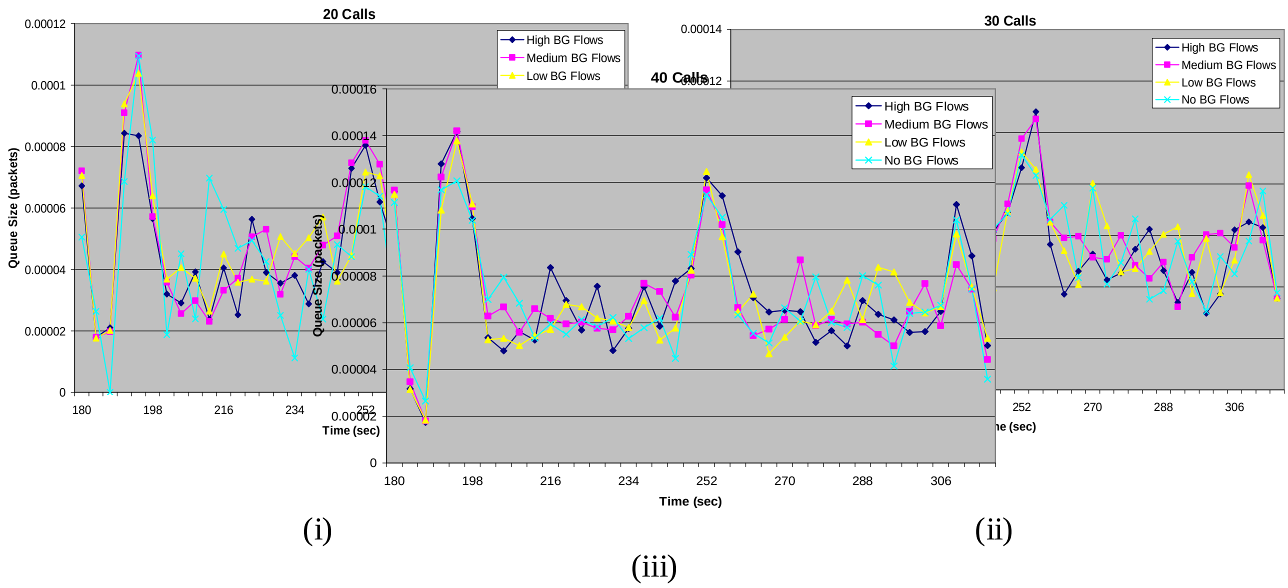


Diagram 68. Average voice queue size under no BG flows, 40 low, medium and high BG flows for: 20 calls (i), 30 calls (ii) and 40 calls (iii)

5.8.4 Conclusion

As the above results show that 40 VoIP calls can be achieved over an ESS environment consisting of 22 routers under 40 background flows of varying demands. With the environment of the calls being placed in an environment that consists of a large number of routers, 22 in this case, means that calls from nodes are placed at different areas of the environment and can be demonstrated from the experienced longer ETE delay. This is demonstrated by diagram 67(i), which shows how 20 calls ETE delay is higher than both 30 and 40 calls scenarios. This inconsistent increase however is not apparent in the average queue size as there is a noticeable and constant increase of 0.00002 when an additional 10 calls are placed in the range of 20 to 40 calls.

5.9 Small ESS: Video Conferencing Calls

The section that follows shows the results of 10, 15 and 20 video conferencing calls placed across the small ESS topology which consists of six routers.

5.9.1 Traffic Received

The average packets per second received by the called nodes under different background flow demands and the number of calls are shown in diagram 69 below. It is clear that under none of the scenarios shown below did the expected 10 packets per second be achieved. Even when calls are placed under perfect network conditions, that of no background flows, diagram 69 shows that it did not impact on network performance in that it did not improve the number of packets received by nodes. As discussed previously through there being more than one WAP, mobile nodes associate with different WAPs as they move within the environment. This therefore means the traffic load is likely to be distributed between the six WAPs in this environment, however as the results show this did not result in increase video conferencing calls being achieved. Thus this low level of performance in receiving packets can be a reflection in the number of dropped bits as show in diagram 72 below.

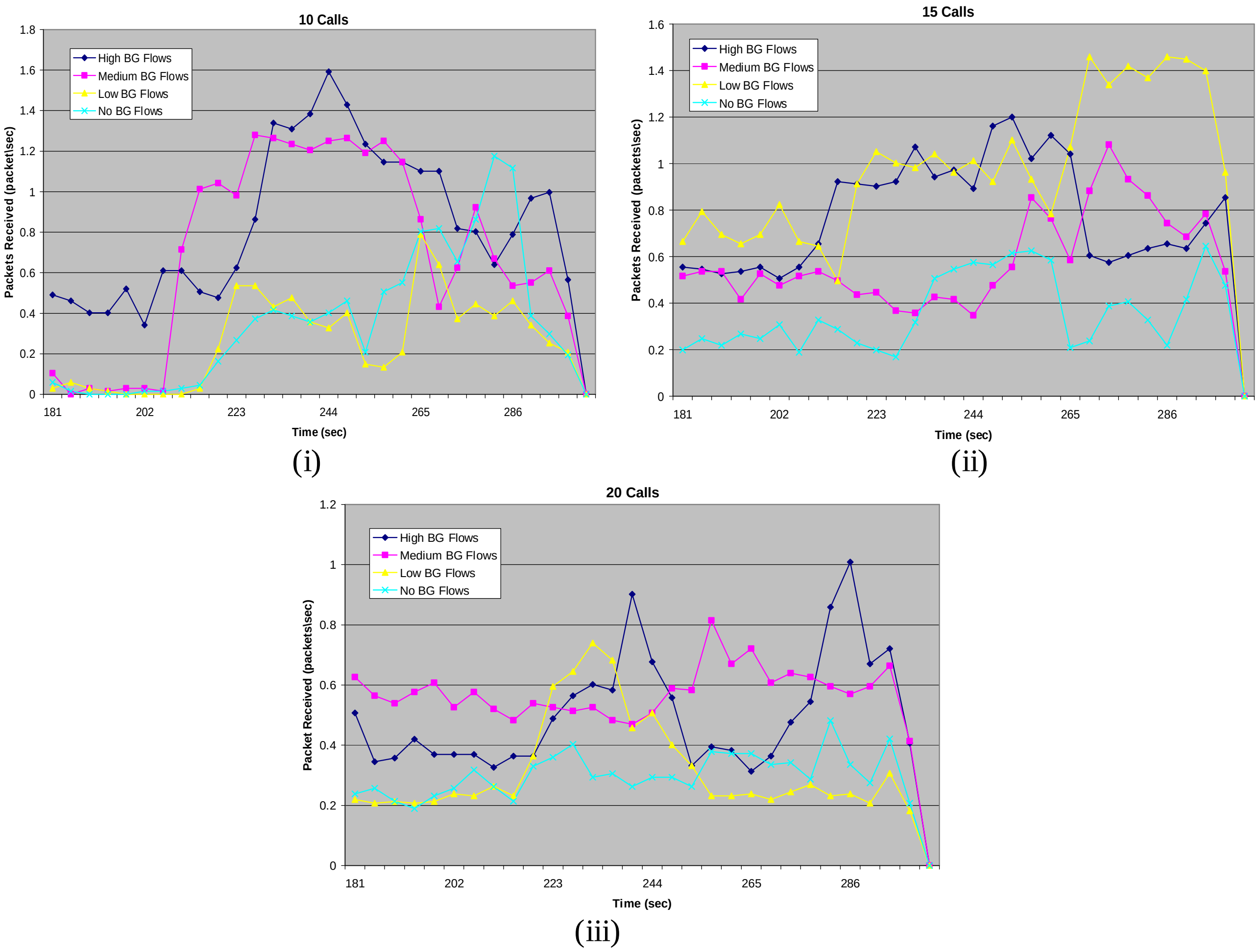


Diagram 69. Average received traffic under no BG flows, 10 low, medium and high BG flows for: 10 calls (i), 15 calls (ii) and 20 calls (iii)

5.9.2 ETE Delay

For the packets received the ETE delay experienced by these packets are shown in diagram 70 below. From the ETE delay it is noticeable that the majority of the scenarios met the ITU recommendation of 150 milliseconds. There is however spikes in ETE delay at specific points for example in diagram 70(i) at the 200 seconds point and 70(ii) at 235 seconds in which when viewed with the results in queue size (diagram 71) and data drop (diagram 72) below, the data indicates that the nodes were located at positions that required a number of hops to reach the destination. Thus indicated due to there not being spikes in the queue size or buffer overflow at these time spots. The change in different background flows demonstrates that ETE delay is relatively unaffected by the change, due to the results of diagram 70(ii) where 'High BG Flows' performance is better for the vast duration of the 120 second call period with a noticeable 0.12 seconds of a difference at the time point 235 seconds. This is further demonstrated in diagram 70(iii) at time point 244 seconds when 'Low BG Flows' and 'High BG Flows' are compared and there is difference of 0.125 seconds between the ETE delay experienced.

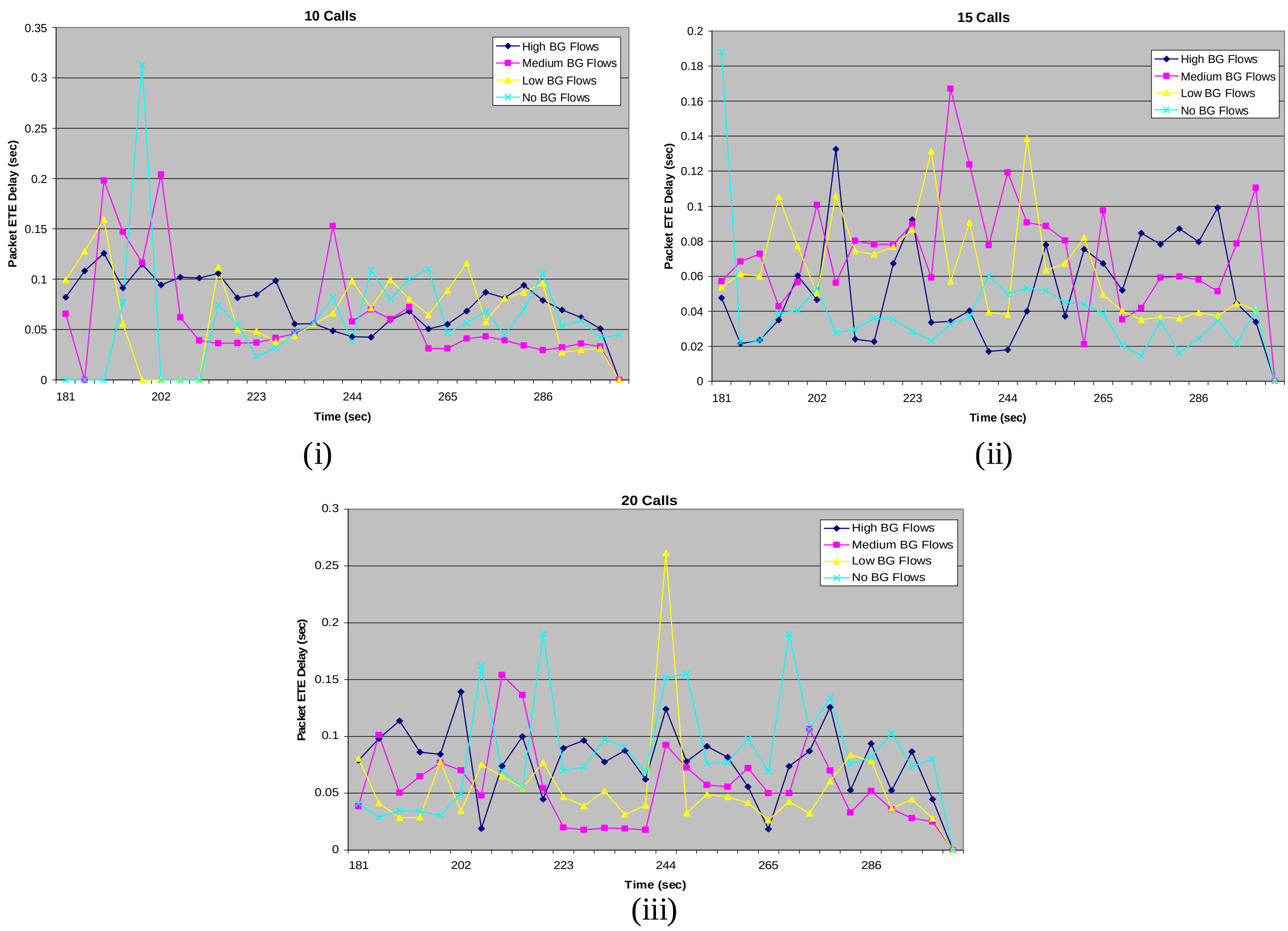
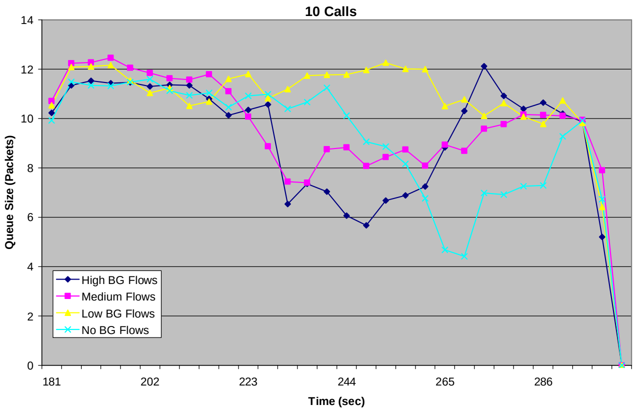


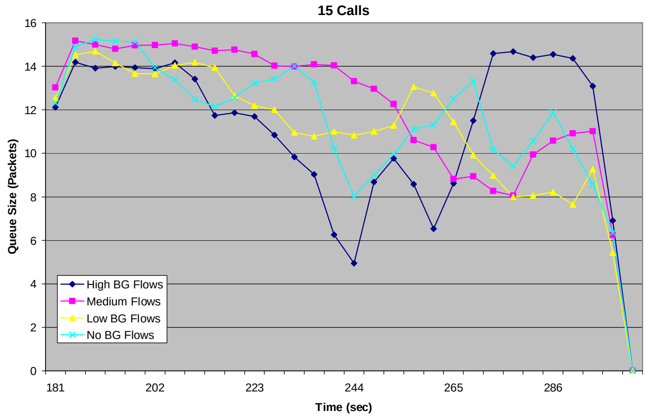
Diagram 70. Average packet ETE delay under no BG flows, 10 low, medium and high BG flows for: 10 calls (i), 15 calls (ii) and 20 calls (iii)

5.9.3 Queue Size

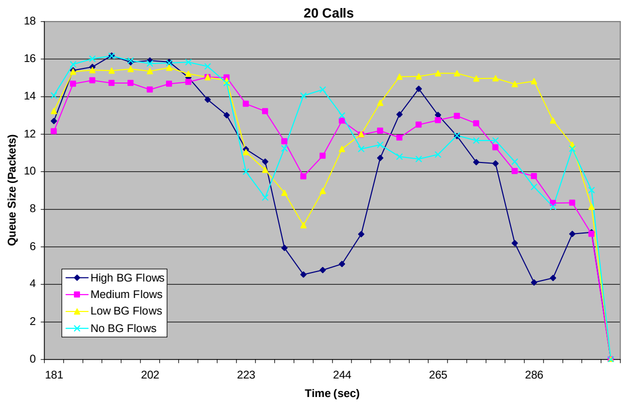
As shown from the previous video conferencing results, it has been demonstrated that video conferencing is an application that relies on queuing (buffering) of packets on devices due to the large number of packets that are generated per seconds. Diagram 71 below shows the average video queue size for all six routers for the 120 second call length when calls start at the 180 seconds. Again the results show that the background traffic demand does not affect the queue length since the traffic is held in a different queue. The results therefore demonstrate how the movement of the nodes and the successful received traffic is reflected by the small queue size. This is illustrated by the highest number of packets being received for 10 calls under high background flow demands (diagram 69(i)) above and 71(i) below during time period 230 and 260 seconds.



(i)



(ii)

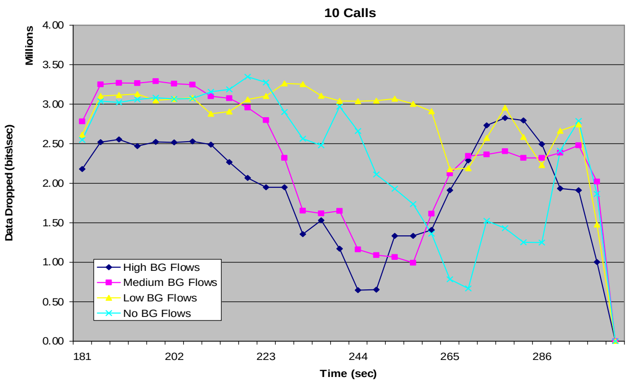


(iii)

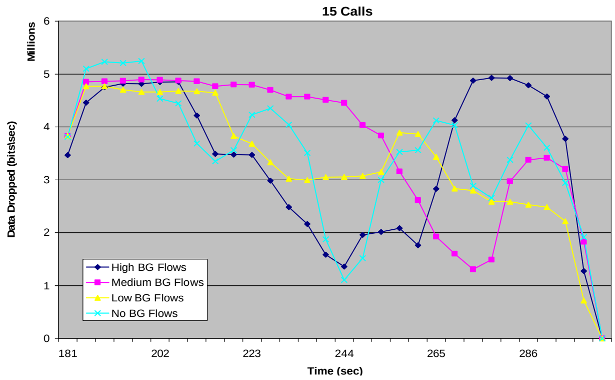
Diagram 71. Average voice queue size under no BG flows, 10 low, medium and high BG flows for: 10 calls (i), 15 calls (ii) and 20 calls (iii)

5.9.4 Video Buffer Overflow

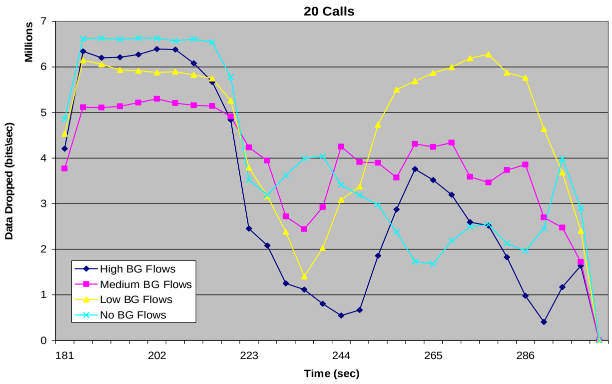
The relation between the packets received, queue size and video buffer overflow is important as the relationship between helps to identify the trends the data shows. Diagram 72 below illustrates the buffer overflow for video traffic in regards to bits dropped per second for 10, 15 and 20 calls under different 10 background flow demands. The results show a reflection of the traffic received in regards to when the number of dropped bits decreases then the packets received as a result increases. This is most noticeable in all three diagrams in regards to 'High BG Flows' in diagram 69 and received traffic in diagram 72 during the time period of 230 and 265 seconds. This drop in the number of bits can be seen to reflect node movement and the distribution of node association to WAP in the environment. During the first 40 seconds of the calls it can be seen that there is a significant difference between 10, 15 and 20 calls with an increase in the number of dropped bits of 1 million as the number of calls increase. Thus as expected since there are additional calls being placed on the network generating more traffic therefore more bits dropped since the router buffers do not change in size.



(i)



(ii)



(iii)

Diagram 72. Average video buffer overflow under no BG flows, 10 low, medium and high BG flows for: 10 calls (i), 15 calls (ii) and 20 calls (iii)

5.9.5 Conclusion

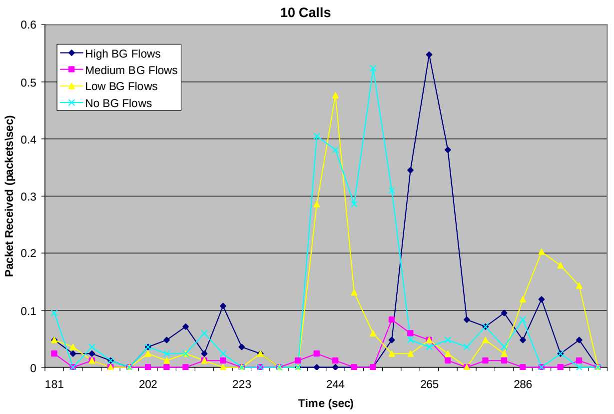
When video conferencing calls are placed across an ESS topology consisting of 6 routers it can be said to be unable to handle 10 calls under perfect network conditions of having no background traffic being sent across the network. It can be said similar to MANET due to ESS having several WAPs that a node can associate with whilst moving within the environment and therefore means that there is no clear or consistent differentiation between scenarios as the data results above demonstrates. This is due to the traffic generated by nodes being distributed between WAPs rather than routed through a single WAP. The data dropped is considerable due to buffer overflow on average dropping more than 6 million bits a second in all scenarios for the three calls implemented and is a reflection the demands video conferencing places on WAPs video buffer.

5.10 ESS Large: Video Conferencing Calls

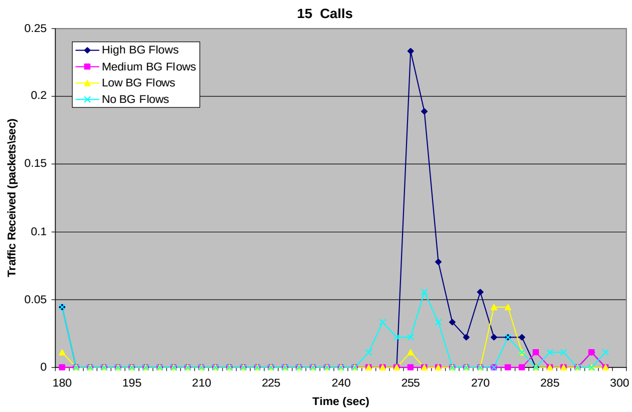
This section demonstrates the results of video conferencing over the large ESS environment which consists of 22 routers.

5.10.1 Traffic Received

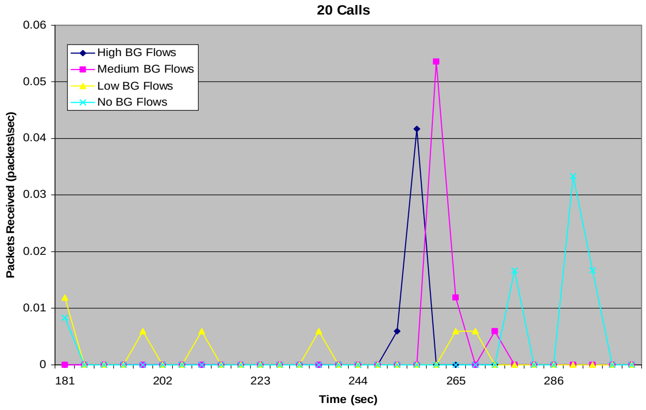
As diagram 73 below shows the average packets received for the duration of the 120 seconds that calls last under different background flows. This reflects the performance of video conferencing up to this point in the report as no calls were successfully achieved under any of the scenarios in diagram 73 below. With on average not even a single packet being received per second it demonstrates the lack of support that ESS to video conferencing calls provides. When compared to ESS small in diagram 69 above it can be seen that there is a drop in the number of packets received and this is a reflection of the additional WAPs and the number of hops that packets potentially have been routed across. Diagram 73(ii) illustrates the impact of mobility over a large ESS environment through packets not being received until 240 seconds in regards to ‘No BG Flows’, which indicates that source and destination nodes moved closer to each other and therefore the packets did not have to be routed across a high number of hops. As shown by ESS small environment the change in background flow demand type has no impact on the performance increase in call capacity, demonstrated in diagram 73(ii) where ‘High BG Flows’ received more packets per second than ‘No BG Flows’ scenario. Thus it can be seen to be the result of the impact of node mobility within the testbed environment.



(i)



(ii)

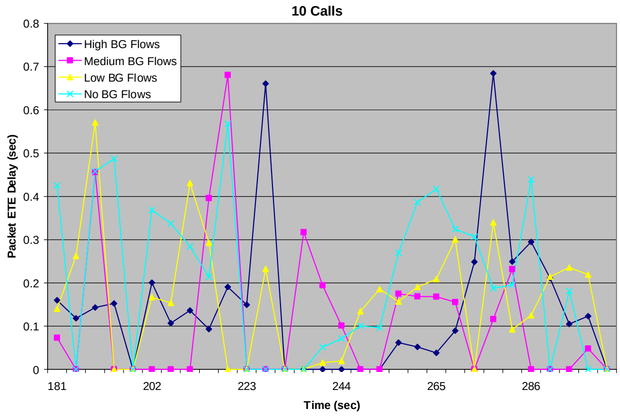


(iii)

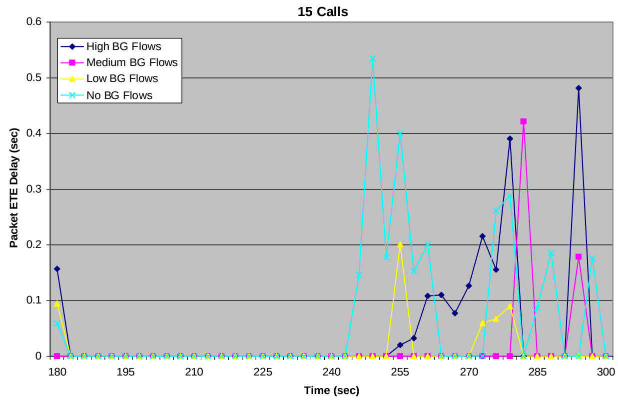
Diagram 73. Average received traffic under no BG flows, 10 low, medium and high BG flows for: 10 calls (i), 15 calls (ii) and 20 calls (iii)

5.10.2 ETE Delay

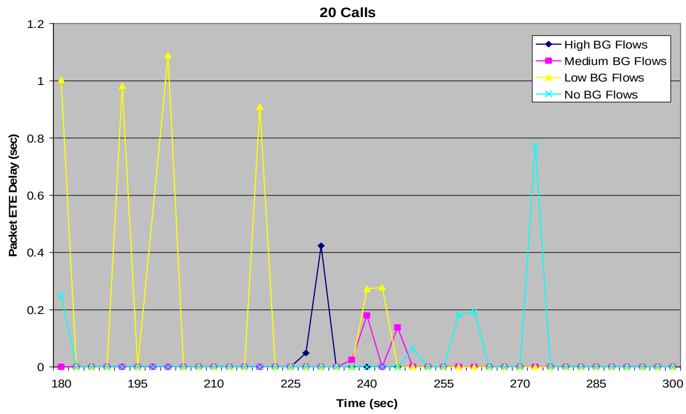
The ETE delay results in regards to 10, 15 and 20 video conferencing calls placed over the large ESS environment with 10 varying background flows is shown below in diagram 74. What is of particular interest is that despite the low number of packets received the ETE delay experienced by large number of packets was above the ITU recommendation of 150 milliseconds. This is well demonstrated in diagram 74(i) for the 'No BG Flows' scenario that for the duration of time between 250 and 286 seconds the delay is above 150 milliseconds and peaks at 420 milliseconds at time point 286 seconds. This high delay can be concluded to be due to the potential number of hops that a packet is sent across the network.



(i)



(ii)

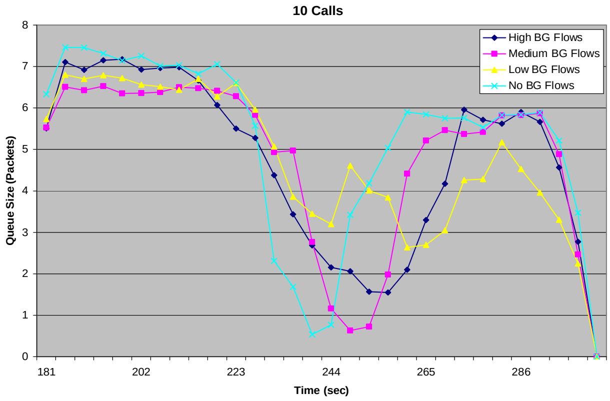


(iii)

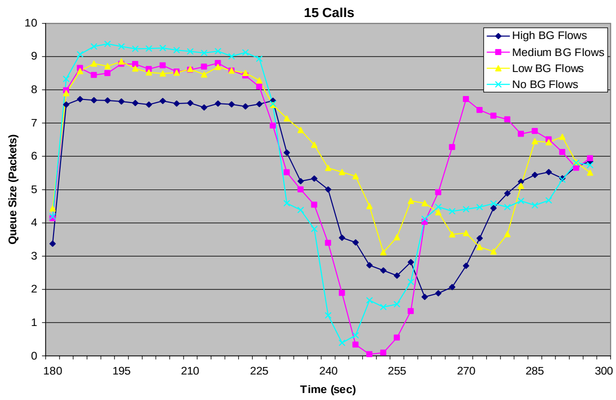
Diagram 74. Average packet ETE delay under no BG, 10 low, medium and high background flows for calls: 10 (i), 15 (ii) and 20 (iii)

5.10.3 Queue Size

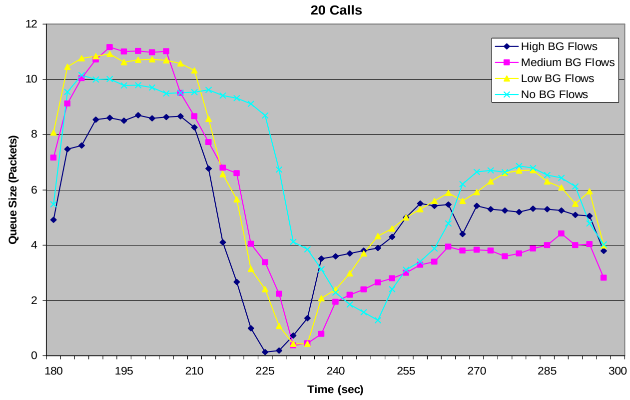
Below in diagram 75, is the average video queue size for the 22 routers that are in the ESS large environment. What is consistent and within all three diagrams of diagram 75 is the average increase of 2 packets per second for the first 40 seconds of the call starting as the number of calls increase by 5. Despite 'Heavy BG Flows' putting more demands on the network it can be seen from diagram 75(ii) and 75(iii) that the queue length was less than when the network was under no background flows. As discussed previously the type of background flow demand placed on the network traffic has no effect on the video queue size since background traffic is stored in a separate queue. The differentiation in performance can therefore be seen to reflect user movement in the environment and association to WAPs as the data collected from buffer overflow discussed below.



(i)



(ii)



(iii)

Diagram 75. Average voice queue size under no BG, 10 low, medium and high background flows for calls: 10 (i), 15 (ii) and 20 (iii)

5.10.4 Video Buffer Overflow

Diagram 76 below demonstrates the video buffer overflow for calls 5, 10 and 15 under 10 varying background flow demands measured in bits per second. What can be seen is that the trends of the scenarios shown in diagram 76 reflect that of the queue size shown in diagram 75 above. As can be seen in the diagrams below when the decrease in the buffer overflow at time point of 240 seconds the packets received by nodes shown in diagram 74 increased. This demonstrates the performance increase when the video buffers are not overflowing.

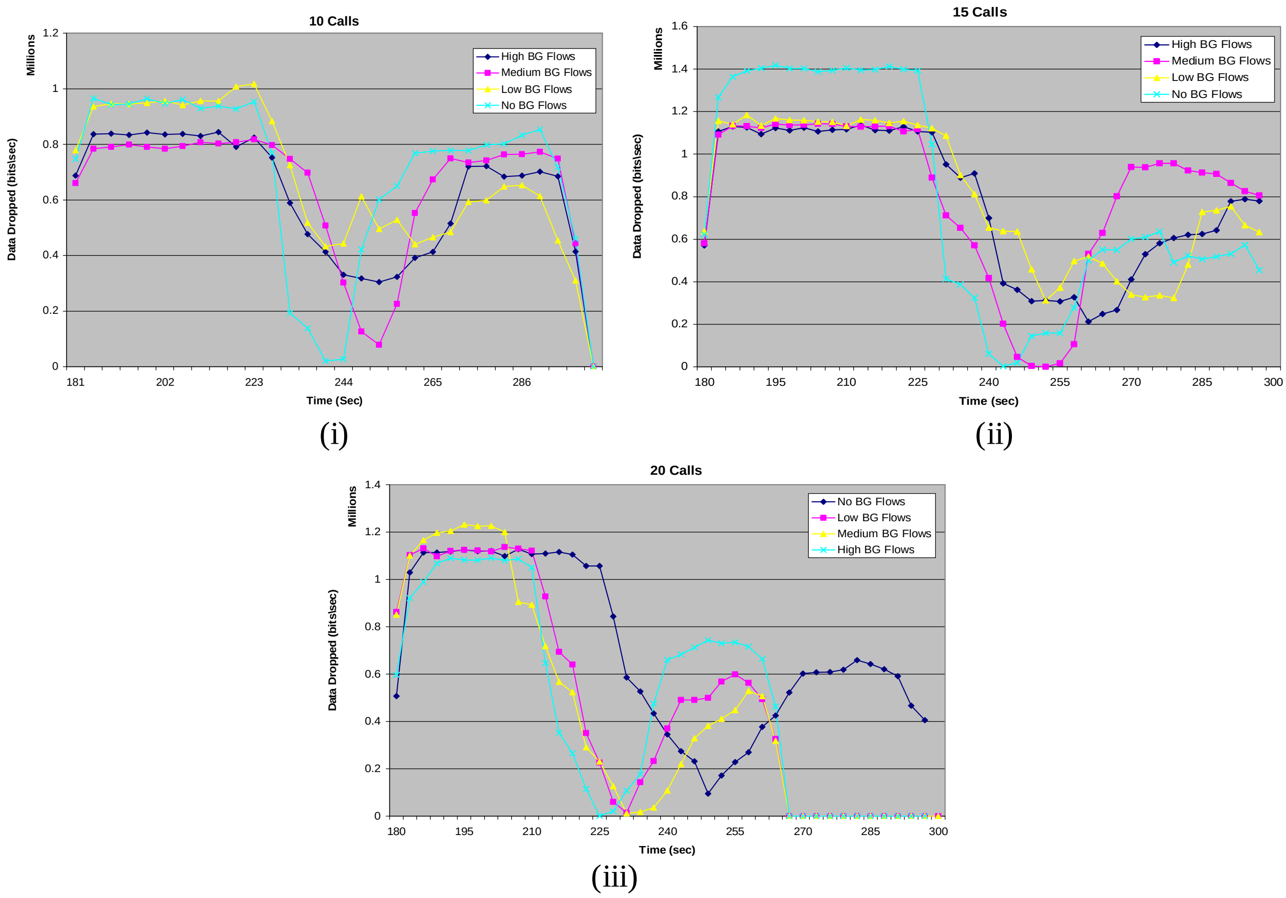


Diagram 76. Average video buffer overflow under no BG, 10 low, medium and high background flows for calls: 10 (i), 15 (ii) and 20 (iii)

5.10.5 Conclusion

As experienced in the small ESS environment it is the case when video conferencing is implemented over a larger ESS environment containing a larger number of WAPs is that it is still unable to successfully deliver video conferencing calls to nodes. The high bit drop rate is less than that experienced in the small ESS environment however this is due to the calls being placed over a larger number of WAPs that may experience no packet loss as they do not receive any video traffic. This therefore reduces the overall average bit rate drop that specific WAPs experienced.

6. Summary and Conclusions

The results from this project have identified the call capacity of VoIP and video conferencing under different background flows over three commonly deployed wireless technologies. The following section considers the project results in relation to the research question and hypothesis set in section 1.2. The findings are then compared to those identified within previous research, as documented within the literature review of this report (section 2). The section will conclude by looking at the limitations associated with this research, due to both its scale and time frame, as well as identifying areas for future research.

6.1 Research Question

Following research into the deployment and popularity of wireless technology, it was identified that the demand for VoIP and video conferencing is extensive. It is therefore logical to conclude that call capacity is an important area and research into this can provide valuable information. On this basis the following research question was derived:

> "What is the call capacity of VoIP and video conferencing when delay, jitter and packet loss are the metrics of QoS for each end to end call when simulated over WAP, MANET & ESS wireless technology environments, which are populated with dynamically moving nodes."

6.2 Research Question Results

The conclusion from the test data results in regards to the research question for WAP, MANET and ESS for VoIP and video conferencing are discussed below.

6.2.1 VoIP

WAP

The results showed that 9 calls can be achieved that meet the ITU recommendation under perfect network conditions, in other words without the existence of background traffic. When background demands are introduced 9 calls can be achieved to the point that 10 medium background flows are implemented. When higher background and more demanding background flows are applied it is not possible to achieve 8 calls that meet the ITU recommendation.

MANET 200 nodes

The dynamic nature of MANET and the low number of mobile nodes for the size of environment is a strong indication of the reasons why no calls were able to be achieved under any of the tested scenarios.

MANET 400 nodes

The introduction of the additional 200 nodes into the same testbed environment of that of 'MANET 200 nodes', provides a significant improvement in the number of calls that were successfully made by approximately, with 25 calls being able to be achieved. However the nature of MANET topology means that achieving this call capacity is not guaranteed as demonstrated by diagram 48, as under 10 calls with low traffic no calls were achieved but under no background, medium and high background flows 10 calls were achieved.

ESS small

With the environment consisting of 6 routers with a wireless routing backbone it can be concluded that 40 calls can be achieved under all scenarios expect that of 20 calls under heavy background flow demands. This result reflects the nature of ESS and it functionality of allowing users to move within the environment and associating with different WAPs, which in this case meant that a single WAP was over populated.

ESS Large

When the ESS topology environment is increased in size to consist of 22 routers it is noted that 40 calls which meet the ITU recommendations is achievable under all scenarios tested.

6.2.2 Video conferencing

WAP

No calls were able to be successfully achieved even when one call under no background traffic was tested. This is due to the voice buffer overflow resulting in millions of bits per second being dropped.

MANET 200 nodes

10, 20 and 30 video conferencing calls were tested across the MANET topology and was unable to achieve a single call receiving 10 packets a seconds for the 120 second call length. What was noticeable is that at points it was possible to receive 10 packets, but this was only for a short period of time, subsequently indicating that node movement impacts on the call performance.

MANET 400 nodes

Unlike the performance increase noticeable in VoIP when an additional 200 nodes are implemented into the environment, in the case of video conferencing there is no performance increase. No calls were able to be achieved and the results demonstrate that the number of calls that were able to establish a connection to there destination caller node was less than that under the same condition of the 200 node MANET environment testbed. Thus highlighting the dynamic nature of MANET and its ability to provide a reliable network performance.

ESS Small

With the network load being spread over potentially 6 routers means that a single WAP is not responsible for delivering all the network traffic. Despite this no calls were successfully achieved with it being noticeable that the introduction of background traffic has no effect on the performance in relation to packets received.

ESS Large

The results demonstrate that video conferencing is not achievable under this specific testbed as on average no calls were able to receive a single whole packet for a single second for the duration of the 120 second call.

6.3 Project Aims

Below is how the project aims, as outlined in section 1.2.1 were achieved.
Number 1: Create three wireless environments that reflect ‘real world’ environments in relation to each wireless technology.

Achieved
This has been achieved by each environment being to the scale that reflects each of the technology deployment and node population. This is demonstrated in the case of the large ESS environment being taken from Cisco literature into the recommended poisoning of WAPs, with the smaller ESS being created from a smaller section of the large ESS. With MANET being a topology deployed where there is no infrastructure an outdoor music festival was selected as users move freely within a large space. The WAP environment reflects a retail shop or warehouse that can be covered by a single WAP.

Number 2: Identify user movement within a ‘real world’ environment.

Achieved
As discussed in section 2.6 achieving user movement within a simulation environment which is representative of the ‘real world’ is on its own an area of research. To replicate as realistic as possible user movement within each typology three movement ranges were assigned to nodes to reflect different movements within the environment. A random distance that nodes could move within the range was also implemented, thus increasing the range of possible node movements.

Number 3: Identify call capacity of VoIP and video conferencing calls over each wireless technology.

Achieved
This was documented and discussed from the primary data gathered from the experiment, as documented in section 2.7 above.

Number 4: Analyse the quality of the calls to identify whether they meet the ITU recommendations.

Achieved
Used as the metric which call quality was evaluated under and produced the call capacity that is documented in aim number 3.

Number 5: What effect does changing the characteristics of the environment have on the call capacity?

Achieved
Through the implementation of 25%, 50%, 75% and 100% of the node population in regards to the number of background flows and having low, medium and high background flow characteristics meant that the environment was significantly tested in relation to environment characteristic changes. Through there being 13 scenarios per number of calls, it provided and is shown in the result section how the call capacity was affected by these environment changes.

Number 6: Identify the call capacity to node population ratio in each environment.

Achieved
By achieving aim 6 means that it can be documented if the wireless technology is able to handle the number of calls to the number of nodes within each environment. It can be stated that in WAP environment up to the point of 10 medium background flows that 90% of calls to node population in regards to VoIP calls can be achieved. Thus the result of 20 nodes in the environment and therefore the maximum number of calls being achievable is 10, since one node is the caller and another node is the receiver. The nature of MANET and the limited call testing further discussed in section 6.7.2 below, it can only be concluded that when there is 400 nodes, 25 calls was achieved therefore the ratio for VoIP is 12.5 %. In the case of ESS small and ESS large topology the ratio proves to be 100 % in both cases. In regards to video conferencing calls no topologies were able to successfully achieve a single call thus the call capacity to node population can not be deduced.

Number 7: Analyse the effect of the additional media stream of video used in video conferencing and how it affects the call capacity in relation to VoIP calls.

Achieved
The effects of implementing video conferencing calls compared to VoIP calls is well documented in this report through the test data presented in the result section. It clearly demonstrated that video conferencing is not capable under the testbed environments in this report by the low levels of traffic received by nodes and the high number of bits dropped due to video buffer overflow.

6.4 Hypotheses Results

The hypothesis documented in section 1.3 were:

- In a WAP environment the call capacity will be VoIP close to 7-8 calls, with video conferencing being less.

Result
From the test data this hypothesis was very close to being matched and thus support existing literature in regards to the call capacity of WAP. The results obtained from this project were that 9 calls to the point when less than 10 or less medium background flows were introduced into the network. This is still higher than the highest documented 8 calls by Geier (2007), thus could be the result of the 802.11g standard having a bit rate of 54Mbits/s. In regards to video conferencing it was not expected that the difference in performance compared to VoIP would be as significant as that which the test data showed. The additional 18240 bits that need to be sent has a noticeable impact when queue size and buffer overflow are examined, and indicates that the WAP needs a larger buffer size when video conferencing is deployed to reduce the number of bits dropped thus increasing the number of packets received.

- In the MANET environment there will be a node capacity that the performance will decrease due to either being too few or too many nodes.

Result
The increase in nodes from 200 to 400 did have a noticeable impact on the performance in regards to VoIP as 25 out of the 30 calls was achieved under several different background flow scenarios. This backups Kleinrock and Silvester (1978) and Royer et al (2001) research that 6-8 nodes is the optimum number of neighbouring nodes a node should have. A limitation of this report is the ability to identify if every node had 6-8 neighbours, as it was not the focus of this study. What was shown is that in the case of VoIP scenarios tested that by increasing the number of nodes in the

environment and therefore increasing the probability of the number of neighbouring nodes a node may have was that it does have a positive impact on the performance of the calls. In regards to video conferencing it contradicts the performance improvement by increasing the number of nodes in the environment. Thus in the 200 node environment there was more calls that established connections compared to that in the 400 node environment, highlighting the dynamic nature of MANET and the ability to provide a reliable network performance over this topology

- The total call capacity in ESS environment will not scale to the number of AP's x calls capacity of a single WAP.

Result
This hypothesis was not tested in its entirety since the VoIP capacity within each environment indicates that more calls would have been able to be implemented if there had been more nodes in the environment. However the node population of the environment reflects the size of the environment and therefore the hypothesis can only be analysed in regards to this factor. What was shown was that in the ESS small environment consisting of 6 routers achieved the maximum possible calls for that environment by 40 calls successfully meeting the ITU standard, thus approximately 6 calls per WAP. In the case of the large ESS environment the scale of calls to WAP is 1.8 calls. Thus the hypothesis can not be proven due to both ESS small and large being able to achieve the maximum number of calls to the number of calls in the environment. It is suggested that from the trend of the data gathered that the calls capacity of a single WAP will scale the call capacity to the number of WAPs there are in the ESS topology, however further research into this is required to allow a conclusive conclusion to be made. In regards to video conferencing no conclusion can be made, as no successful calls were achieved which was also the result when tested in a single WAP environment.

6.5 Relation to Previous Work in this Area

The findings of the report are now compared to those identified within previous research discussed in the literature review.

WAP

The most interesting previous research was David and Keegan (2006) paper that concluded that the introduction of background traffic with packet sized above 256 bytes impacts on performance by only 60% of the AP delay being below 10ms. From the results gathered from this experiment it can be documented that the introduction of background traffic sized 512bytes had no significant impact on the ETE of calls. For example in diagram 32 where the ETE delay experienced was level in performance to low and medium background demand traffic flows. This delay decrease may not be experienced due to the higher bit rate that 802.11g supports over 802.11b standard but also to the implementation of the 802.11e standard that priorities time sensitive traffic such as voice and video traffic being sent before background traffic. The study by Coyle et al (2006) study did not implement interactive video conferencing but rather streaming to static users in a football stadium being able support this service to 10 users. This identifies that making the video conferencing one way has a significant impact as it reduces traffic load as it is only one way traffic being passed through the router because of streaming. If node/user mobility is introduced it could also impact on performance seen by this study and be a reason for the performance identified from this report since the WAP cannot be installed in a position that optimises nodes positioning compared to when nodes are static.

MANET

A study in 2005 by Ceo et al identified when Ad Hoc technology is implemented using static nodes that 8 video conferencing calls was possible. However no comparison or data gathered from this report experiment using MANET, which uses a very similar technology than Ad Hoc was able to support Ceo et al results. What could not be supported was Houle et al (2006) view that VoIP could not be deployed over an infrastructureless topology, as 25 calls were achieved in this report in the 400 node scenarios however when the performance of VoIP over 200 node scenarios Houle et al (2006)view was supported. The problem with MANET as documented throughout this report is that its dynamic nature means that the same scenario run produces different results if random mobility of nodes is implemented since the performance of the network is dictated by the position of nodes within the environment. As discussed Kleinrock and Silvester (1978) and Royer et al (2001) identified the optimum number of neighbours a node should have within the environment to increase the probability that a route to the destination node being found. It is outside the scope of this report to identify how many neighbours a node had at a specific point in time, however in the case of VoIP the increase in the number of moving nodes within the environment did improve the number of successful VoIP calls achieved. Further research into identifying how many nodes is discussed in section 6.7.2 in regards to future research that could lead on from this project.

ESS

Van Geyn et al (2009) ESS research identified that as the number of hops a call had to be delivered over to reach the destination node the call capacity drops by 25 % when two hops to the destination is compared to a single hop route. This decrease continues leading to 5 hops supporting a single call. What can be concluded from the results gathered was that the increase in environment size which contains 3 hops (ESS small) and 6 hops (ESS large) diameters was that there was no noticeable effect on the call capacity experienced. A limitation of this study is that movement of the nodes are not recorded therefore are unable to track the location of where calls were placed across the network as it was not in the interest of this study.

6.6 Irregular Data Results

As documented through the results it has been noted that there have been irregular increases in received packets during the time period of 240 and 300 seconds. It was noticeable under different tested environments and scenario's indicating it is not a miss configuration in a single scenario as all individual G.711 VoIP profiles and any duplication of VoIP demands were checked to be correct. The software manufacturer (OPNET) have been contacted as of when the problem was identified however a reply back has yet to be received back as of the 06/04/2010 and therefore still inconclusive on the reason for this irregular increase in received traffic.

6.7 Strengths and Limitations of Study

Despite limitations existing in relation to this report, due to its scale and time frame, a number of areas were considered to be carried out successfully.

6.7.1 Strengths

The number and broad range of results shown in section 5 demonstrates the wide range of data that was collected during the execution of the experiment. Through collecting this number of statistics, which were not all published due to time limitations, the results that were documented were able to be supported with additional statistics to provide further information into why the documented results were produced. This is demonstrated from the buffer overflow and queue size statistics gathered in WAP and ESS environment, which highlights why the received packets did not meet the expected number of packets through queue size and buffer overflow rather than putting the results down to the vulnerable nature of wireless technology. In the case of MANET and ESS topologies, which there topology are specific in regards to number of nodes and WAPs respectively, was the reason why both these topologies were tested under two different environment scenarios. The MANET topology increased the number of nodes from 200 to 400 nodes and ESS increasing the size of the environment to implement 22 WAPs. The results gathered meant a noticeable increase in successful calls was achieved in regards to increasing the number of nodes in the MANET environment but also illustrated its dynamic nature and lack of performance reliability as illustrated in diagram 48.

6.7.2 Limitations

As stated at the start of this section, the scale and time frame to carry out this project meant that there were limitations associated with the project. A study that implements mobile nodes is time sensitive due to the time it takes for each scenario to run, thus resulted in fewer runs per scenario due to the time scale of the project. Subsequently resulting in a reduction of the range of data captured when there are potentially millions of different node movement scenarios directly affecting the test data that could be collected and analysed. In the case of the report each scenario was run 20 times as recommended, however this recommendation is not specific to when mobility is implemented. This resulted in similar data trends between scenarios being obtained, as illustrated in diagram 76 indicating the node mobility followed a similar patterns. Through running each scenario a higher number of times would increase the probability of obtaining the same result again.

A direct impact on time and the number of runs per scenario was the availability of the required computer equipment necessary to run the simulations, as well as the time it takes to run the scenarios. OPNET software which was used to carry out this project experiment is an application that requires high CPU and memory resources. This meant that in order to have the range of test data that this report collected the number of runs had to be limited to 20, as in the case of the 400 node MANET environment with 30 calls and 400 high background traffic it took on average five hours to complete a single run.

Another limitation existing is from the test data gathered in the case of the ESS environments was that no call capacity in regards to VoIP was found when calls no longer met the ITU recommendations. This was due to there being no free nodes left in the environment to make calls, as all nodes in the environment were either the caller or the called party. However as stated in ‘Project Aim 1’ in section 1.2.1, it was the objective of this project to create realistic ‘real world’ environments not only in size but also with regards to node population. Thus it is argued that this report has found the capacity of the ESS environments since the environment was placed under the maximum number of calls that an environment of this size and node density would have in the ‘real world’.

6.8 Future Work

Although results from the test data gathered have been similar to those previously documented, they have also called the results of other studies into question, as discussed previously in the case of ESS environment. Van Geyn et al (2009) hop count decrease in calls was not noticeable in the data gathered in this report, but nor can it be disproven as hop count was not recorded in this report. Main areas of research that could be further studied as a result of the conclusion of this report is in the area of buffer size, node mobility and further creation of realistic ‘real world’ environments.

Noticeable in WAP and ESS environments was that when video conferencing was the video buffer size had a direct impact on the video buffer overflowing resulting in drop bits. In the case of this report the buffer size was not changed and was set to the software default 256000 bits, which from the results indicates that it was not high enough. Further work into how increasing this buffer size impacts the performance of video conferencing calls would ultimately result in a range of buffer sizes being identified that can be used to identify the buffer size required to the number of supported calls. Before this due to the large amount of test data collected further analysis of the data would be recommended to further analyze and compare results between scenarios in regards to the number of background flows to call capacity and background flow demands to call capacity. Through doing this it would aid in being able to further focus in a particular aspect in the areas discussed below.

As documented in the limitations of the project and discussed within the literature review section in regards node mobility, it is an area that is widely being researched and developed. As discussed previous and from the results shown particularly in regards to MANET scenarios is because node movement has a direct impact on the performance of the network and therefore the test data produced. Carrying on from the conclusions of this report is that investigating node mobility and investigating mobility patterns specific to common wireless deployment scenario that are currently gaining in popularity for example metropolitan networks. Through developing node patterns that encompass user movement and ability to change there movement as a result of an external factor such as someone else walking towards them or a gap in the traffic is the challenge in developing an accurate mobility pattern. By achieving this would not only impact on studies on call capacity, but also other research areas that could implement the mobility pattern, for example GPS.

Building on node mobility is the creation of realistic environments which this project has tried to achieve but is limited to only environment size on node population, without the consideration of obstacles such as desk, chairs and walls. Incorporating these elements into the environment would potentially have a significant impact on the performance experienced since as described in section 2.1.2 different obstacles reflect wireless signals differently thus affecting the signal received by nodes and therefore impacting on the call quality leading to a lower call capacity. Further research into call capacity incorporating these elements could then be compared to the data results gathered from this report and identify if the same size of environment containing obstacles impacts on call performance.

6.9 Conclusions

The completion of this report has identified in relation to the tested wireless environments for WAP, MANET and ESS the call capacity for VoIP and video conferencing. It can be concluded that in all wireless technologies that VoIP is capable of being deployed over them with the introduction of background traffic having a noticeable impact on the call capacity under particular scenarios. Video conferencing and the high traffic demands that a single call generates results in it not being able to be supported over any wireless technology tested. It is thus clear from the results that the size of buffer used by WAPs is unable to buffer the required amount of data and why future research into this area is recommended.

The findings of this report would be of interest to a range of individuals within the academic community due to the report covering VoIP, video conferencing, wireless technologies and mobility patterns. Beyond the academic community this report could be used by enterprises considering the deployment of wireless communication within there business to provide VoIP or video conferencing thus allowing them to identify if there current network demands can be placed onto a wireless network.

It can therefore be concluded that from this report that despite having limitations it is still a credible experimental based evaluation report that investigates if the current trend of deploying LAN based multimedia applications such as VoIP and video conferencing can be achieved over wireless technology. The use of the ITU recommendation metric standards meant that the data gathered was evaluated against a globally recognised and industry used recommendation thus a conclusion of identifying call capacity could be made. Through the execution of the project being logical and thorough resulted in a substantial amount of test data being collected with it only being possible to show the key findings. What is clear is that VoIP by its nature of higher frequent small sized packets compared to video conferencing and its less frequent but lager packet size ultimately resulted in VoIP calls being achievable over all three wireless topologies, whilst video conferencing failed to achieved a single call under the same scenarios and environment that VoIP was tested in.