An analysis of voice over Internet Protocol in wireless mesh networks

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Abstract

This thesis presents an analysis of the impact of node mobility on the quality of service for voice over Internet Protocol in wireless mesh networks. Voice traffic was simulated on such a mesh network to analyze the following performance metrics: delay, jitter, packet loss and throughput. Wireless mesh networks present interesting characteristics such as multi-hop routing, node mobility, and variable coverage that can impact on quality of service. A reasonable deployment scenario for a small organizational network, for either urban or rural deployment, is considered with three wireless mesh network scenarios, each with 26 mesh nodes. In the first scenario, all mesh nodes are stationary. In the second scenario, 10 nodes are mobile and 16 nodes are stationary. Finally, in the third scenario, all mesh nodes are mobile. The mesh nodes are simulated to move at a walking speed of 1.3m per second. The results show that node mobility can increase packet loss, delay, and jitter. However, the results also show that wireless mesh networks can provide acceptable quality of service, providing that there is little or no background traffic generated by other applications. In particular, the results demonstrate that jitter across all scenarios remains within human-acceptable tolerances. It is therefore recommended that voice over Internet Protocol implementations on wireless mesh networks with background traffic be supported by quality of service standards; otherwise they can lead to service delivery failures. On the other hand, voice-only mesh networks, even with mobile nodes, offer an attractive alternative voice over Internet Protocol platform.

Categories and Subject Descriptors

C.2.1 [**Computer-Communication Networks**]: Network Architecture and Design - *Wireless Communication;* D.2.8 [**Metrics**]: Performance Measures; C.2.2 [**Network Protocols**]: Routing Protocols/*Wireless Mesh Network*

Declaration

I, MOHAMMAD TARIQ MEERAN, declare that this thesis "*An analysis of voice over Internet Protocol in wireless mesh networks*" is my own work, that it has not been submitted before for any degree or assessment at any other university, and that all the sources I have used or quoted have been indicated and acknowledged by means of complete references.

Signature: . . . . Date: 20 December 2011
Mohammad Tariq Meeran

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Contents

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List of acronyms

BATMAN	**B**etter **A**pproach **to** **M**obile **A**d-hoc **N**etworking
DSDV	**D**estination-**S**equenced **D**istance **V**ector routing **p**rotocol
DSR	**D**ynamic **S**ource **R**outing
GodRP	**God** **R**outing **P**rotocol
IEEE	**I**nstitute of **E**lectrical and **E**lectronic **E**ngineering
IPD	**I**nter **P**acket **D**elay
LAN	**L**ocal **A**rea **N**etwork
MAC	**M**edia **A**ccess **C**ontrol
MANET	**M**obile **A**d-hoc **Net**work
NCTUns	**N**ational **C**hiao **T**ung **U**niversity **N**etwork **S**imulator
NS2	**N**etwork **S**imulator 2
OLSR	**O**ptimized **L**ink **S**tate **R**outing **P**rotocol
QoS	**Q**uality **o**f **S**ervice
RTCP	**R**eceive **T**ransmission **C**ontrol **P**rotocol
RTG	**R**eceive **T**raffic **G**rapher
RTP	**R**eal-time **T**ransport **P**rotocol
RTT	**R**ound **T**rip **T**ime
STA	**Sta**tion
STCP	**S**end **T**ransmission **C**ontrol **P**rotocol
STG	**S**end **T**raffic **G**rapher
TCP	**T**ransmission **C**ontrol **P**rotocol
UDP	**U**ser **D**atagram **P**rotocol
VoIP	**V**oice **o**ver **I**nternet **P**rotocol
Wi-Fi	**W**ireless **F**idelity
WMN	**W**ireless **M**esh **N**etwork
WSN	**W**ireless **S**ensor **N**etwork

1 Introduction

This thesis presents an analysis of voice over Internet Protocol (VoIP) applications' quality of service (QoS) issues on wireless mesh networks (WMNs). The unique characteristics of WMNs can cause VoIP applications to have QoS problems. Studies show that WMNs' unique characteristics are manifested by combining the features of mobile ad-hoc networks (MANETs), wireless sensor networks (WSNs) and cellular technologies [25] [51], but these features cause QoS challenges for VoIP implementations. VoIP applications also have their own characteristics, which include sensitivity to delay, jitter, packet loss and use of small packets. Each one of these characteristics is a QoS factor for any given VoIP application's success. For example, if the delay, jitter and packet loss rate are not matching physical performance thresholds or human-acceptable tolerances, then VoIP QoS will be affected negatively. This can result in user dissatisfaction and complaints about the quality of service level. The research described in this thesis focuses on studying VoIP applications' QoS by exploring how this type of traffic is affected by different WMN scenarios, particularly when some or all nodes are mobile. Many WMN scenarios exist today. These include mesh networks with no mobility, mesh networks with limited node mobility and mesh networks with full node mobility. The mobility speed can also vary in the latter two scenarios, based on the type of equipment and the purpose for which it is being used. A mesh node can have a slow mobility, like pedestrian users, medium speed, like a bicycle or yacht, and faster speeds like motorized vehicles, e.g. motorcycles or trains.

1.1 Background

A WMN is a form of wireless LAN (Local Area Network) which is considered to be a type of ad-hoc network, while sharing characteristics of mobile ad-hoc networks, WSNs and cellular networks. WMNs can be formed by a mix of wireless clients, wireless access points and wireless routers (gateways). The wireless clients/access points/routers can be stationary or mobile and can be implemented alone or mixed with other devices to form WMNs. They provide multiple and redundant paths (routes) for each other. WMNs are multi-hop, self-healing, self-organizing networks with dynamic topologies, using purpose-built routing protocols. WMNs are widely deployed in areas where building wired infrastructure requires a considerable amount of time and money like the emergency relief hotspots, rural areas, battlefields, education etc., in order to support various applications and services, like web browsing, e-mail, file transfer (non real-time), voice and video (real-time). Non real-time services are not greatly affected by delay and jitter that are produced as a result of high loads, low bandwidth, longer distances and mobility, but real-time traffic is affected, since it requires lower latency, lower jitter, affordable packet loss rate and better throughput. WMNs provide better coverage, throughput, redundancy and fault tolerance, but there are some QoS problems, due to WMNs' dynamic and complex topologies, multi-hop nature, node mobility, medium usage routing/relaying capabilities, bandwidth allocation and traffic prioritization for real-time traffic, which require affordable delay, jitter and packet loss.

The IEEE (Institute of Electrical and Electronic Engineering) 802.11’s standard for WMNs considers the mesh nodes as part of the network infrastructure, whether stationary or mobile [51]. In this research WMNs’ formations will only be by wireless mesh clients. This research does not consider wireless access points and wireless routers to be part of the mesh design and topologies. Studies show that WMNs introduce more delay, jitter and packet loss and they may be causing problems for the VoIP applications [40]. Besides, WMNs do not share a single point of failure. If one node fails for any reason, another node is selected instead. For example, Figure 1-1 shows a wireless mesh node where node A can reach node D, using any of the following available routes:

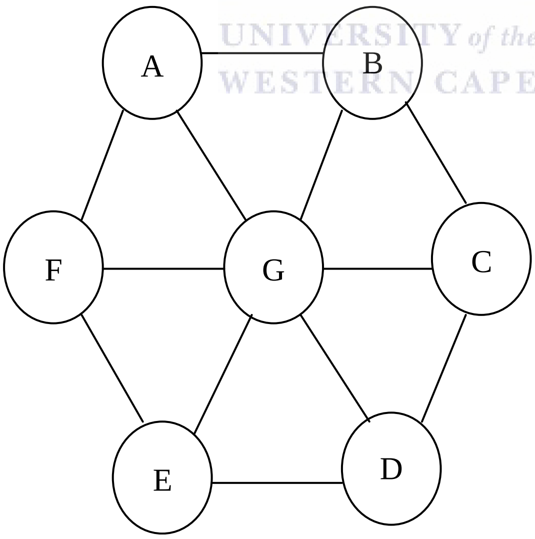


Figure 1-1 *Wireless mesh network example*

1. B→C	2. B→C→G	3. B→C→G→E	4. B→C→G→F→E
5. G	6. G→C	7. G→B→C	8. G→E
9. G→F→E	10. F→G	11. F→G→C	12. F→G→B→C
13. F→E	14. F→E→G	15. F→E→G→C	16. F→E→G→B→C

As the above figure shows, there are many routes that can be used by each mesh node. Normal wireless networks and routing protocols do not allow mesh nodes to discover and use these many routes. Therefore WMNs use specific routing protocols to make use of these many routes. The wireless mesh nodes must be equipped with wireless mesh protocols and be aware of the topology. They also need to support wireless mesh routing protocols in order to build mesh routing tables, perform load balancing and be able to discover redundant routes and use them if one route or a next hop is not available.

QoS has several meanings. ITU-T defines QoS as "the collective effect of service performance which determine the degree of satisfaction of a user of the service."[1] One definition relates QoS to the mechanism or the technology, which has the power to prioritize data flows based on the application requirements and user satisfaction. QoS is application dependent and requires different levels to satisfy users' needs. In other words, QoS refers to control mechanisms that can provide different priority to different users or data flows, or guarantee a certain level of performance to a data flow in accordance with requests from the application program. Other sources offer the following as a definition: ‘‘QoS represents the set of those quantitative and qualitative characteristics of a distributed multimedia system necessary to achieve the required functionality of an application’’ [9].

The concepts of **delay and latency** are similar concepts in the field of telecommunication. In this research, the term delay is used instead of latency. Delay is defined as the measurement of time between the moment that something is initiated and the moment that its effects begin. In communications it can be measured either as a one-way delay or round-trip-delay. One-way delay is the time taken from when the source sends a packet, until the destination receives it. Round-trip-delay is the time taken when the source sends the packets and the destination acknowledges the reception back to the source. Delay is usually a problem in packet switched networks (connectionless networks). It is a general problem in the telecommunication field. Usually satellite links introduce more delay than wired links. This typically affects real-time traffic, like voice and video in slow speed and congested network connections [49]. Studies suggest that the acceptable delay for voice over Internet Protocol (IP) traffic is 100-150ms [24] [27]. Other studies suggest that acceptable delay for voice over IP is 200ms [49]. The most appropriate way to decrease delay, is reserving bandwidth among the routing and switching nodes that may cause delay between the sender and the receiver, or prioritizing the traffic by QoS methods.

[1] *ITU-T Rec. G.1000 (11/2001) Communications Quality of Service: A framework and approach*

Jitter in the field of telecommunication is used to describe the time difference and intervals between the pulses that are transmitted successfully. It can also be quantified by all time varying signals between source and destination. In packet switched networks, jitter is usually considered a problem. Packet switched networks are designed in such a way that packets can be routed using different paths in the network. Therefore, packets may arrive at different time intervals, due to queuing delay variation at each node [27], which causes problems. Studies show that an acceptable jitter rate between the source and destination communication point is less than 100 ms [38] [48]. If it is more than 100ms, it should be reduced. Usually to solve the problem with jitter, a ‘’jitter buffer’’ is used. It is a small buffer which receives the packets and then transmits them to the receiver with a small delay. If a packet is not in the buffer, it means that the packet is lost or has not arrived, and if it won’t harm the communication, it will not be taken into account. If the size of the jitter buffer is increased, the delay time will be increased, and less packet loss might occur. If the jitter buffer size is decreased, it means less delay, but more packet loss.

Packet loss is defined as the loss of packets (portion of data) during the communication between the stations over a computer network, as shown in Figure 1-2. There are several factors that can cause packet loss. They include degradation of signal, congestions on the network links, low signal quality, corrupted packets, faulty hardware, etc. Packet losses of more than the tolerable rate can cause problems of session disconnections between the two communicating stations. Real-time traffic, like voice and video can be greatly affected by a high packet loss rate. Acceptable packet loss rate for VoIP is considered to be less than 5% of a whole conversation [3].

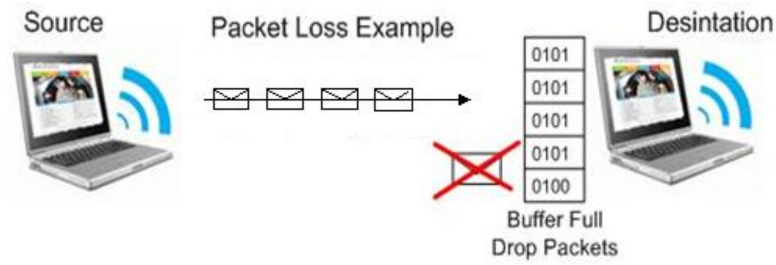


Figure 1-2 *Packet loss*

Throughput is the average amount of data that is successfully delivered over a communication link. The maximum throughput is less than or equal to the amount of digital bandwidth. It is measured by bits/bytes per seconds [55]. Throughput for VoIP traffic varies based on the type of codec being used and the amount of compression applied. A common G.729A codec, with 8 bits compression, uses 32 kbps throughput [50], but if the size of the VoIP payload increases or decreases, the amount of throughput will change accordingly.

Real-time traffic, as shown in Figure 1-3, refers to data flows within the network that have to be transmitted and received almost at the moment that they are generated. They are usually associated with deadlines after which the data will be considered unusable. In this research, the term real-time traffic refers to the voice and video traffic.

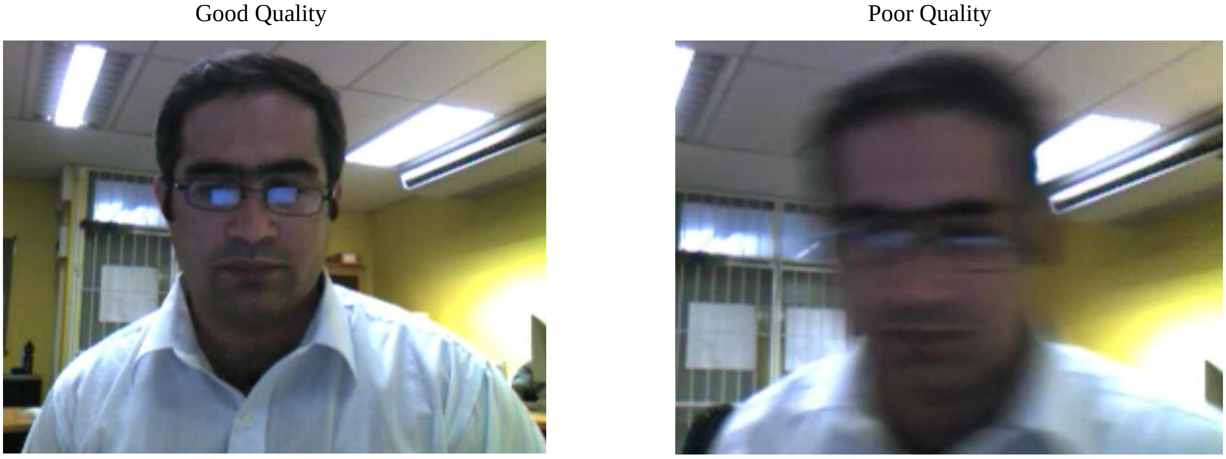


Figure 1-3 *Real-time application video example*

VoIP is the usage of data networks to transport voice over the Internet or internal networks. VoIP mainly provides low cost calls compared to the phone system, using a circuit switched technology [14]. VoIP is supported by several protocols and it has its own set of characteristics which are not common in other types of applications. These include the use of Real-time Transport Protocol (RTP), User Datagram Protocol (UDP), IP, small packets and big packet headers. VoIP can use several types of codec to provide poor, average or high QoS, depending on the type of connection and technology available. VoIP is considered to be sensitive to delay, jitter and packet loss.

1.2 Research question and overall approach

The usage of VoIP applications is becoming more popular daily. VoIP implementation over fiber, copper and wireless networks are becoming more common, but studies show the VoIP applications are having QoS issues in WMNs. Therefore, VoIP implementations in WMNs require more research in order to discover the reason that lead to QoS issues and affect the VoIP QoS factors. This research mainly focuses on WMN scenarios and simulating mesh nodes in stationary and mobile modes. The aim is to discover how and which VoIP QoS factors are affected. The focus of this thesis is to answer the following research question, which will be discussed in detail in Chapter 3: **How is VoIP QoS affected by WMNs' node mobility?**

In order to conduct this research, the NCTUns 6.0 (National Chiao Tung University Network Simulator) was chosen. Three wireless mesh scenarios were designed. The no mobility scenario was designed to simulate mesh nodes in stationary position, the limited mobility scenario was designed to simulate 10 mobile nodes and 16 stationary nodes and the full mobility scenario was designed to simulate all mesh nodes moving. Node movement was configured with the walking speed of 1.3m/sec. In each scenario, two traffic profiles were tested. Traffic generation tools were used to generate traffic. Profile 1 was scripted to generate only VoIP traffic; profile 2 was scripted to generate VoIP and non-VoIP traffic.

1.3 Thesis outline

In order to present how VoIP quality is affected by WMNs' node mobility, this thesis is structured into 5 chapters. The introductory chapter discussed the main focus of the thesis, introduced some of the applicable terms and summarized the research approach.

Chapter 2 presents the related work on the WMNs and VoIP QoS domains. Several types of mesh networks and WMNs are introduced. VoIP QoS and its characteristics and factors follow a discussion of WMN characteristics, formations and types. Finally, the implementation of VoIP applications on WMNs is presented.

Chapter 3 presents the methods, research question, methodological approach, simulation environment, WMN topologies, mesh routing protocols, scenarios and data collection method. It also focuses on VoIP and non-VoIP traffic profiles, simulation software usage, traffic generation tools and commands for simulating WMNs and VoIP traffic.

Chapter 4 presents the analysis of results, with the main focus on how VoIP traffic is affected by mesh node mobility. It analyzes each QoS factor, like delay, jitter, packet loss and throughput separately, considering the WMN scenarios and traffic profiles. Finally, all the scenarios and traffic profiles are compared and the analysis is presented.

Chapter 5 summarizes the research and discusses the research limitations, along with some recommendations and future work.

Appendix **A** presents an unpublished 5-page draft of a paper in SATNAC format. This paper has a co-author, as mentioned in the paper heading, but this thesis is the sole work of the thesis author.

2 Related work

Many researchers have studied WMNs and its effect on VoIP applications. The studies show that WMNs and VoIP applications have some interesting and unique characteristics compared to other types of networks and applications. The standards developed to address the QoS issues in wired and wireless networks do not directly apply to WMNs, because these networks operate differently than other types of networks. Since WMNs are considered a newly-emerged types of wireless networks, it is still an open and active research field for finding ways and solutions to solve the associated problems and provide better service for end users. This chapter is structured to discuss the related work in the WMN and VoIP QoS domain. In Section 2.1 mesh network types are presented in general. Section 2.2 presents WMNs usage types, characteristics and formation. Section 2.3 presents VoIP QoS special characteristics and VoIP QoS critical metrics. Section 2.4 discusses the WMNs and VoIP implementations, with more focus on researchers' efforts and findings in problems associated with VoIP application in WMNs. This section also discusses the measuring VoIP QoS factors in WMNs. Finally, Section 2.5 presents a summary of related work of the research community.

2.1 Mesh networks

Mesh networks are usually formed to eliminate the single point of failures, to provide redundant paths, to be able to self-heal in case of small and big scale outages and to build reliable networks. Many types of network technologies can form mesh networks. Today the use of fiber optic, copper and WMNs are common. Fiber optic mesh networks are built to form intra-building, intra-campus, intra-city, intra-country, regional and intercontinental links. These links form the backbone of the modern networks, as well as the Internet, and are usually used to connect core switches and routers. Copper mesh networks are commonly used in intra building LANs to connect switches and routers. WMNs can be used to connect mobile users, intra building LANs, intra campus LANs and even intra city and country networks. WMNs can connect to user equipment, like mobile phones, laptops, PDAs, desktops and infrastructural equipment, like access points and wireless routers. By considering the mesh network formation and its connectivity to different types of devices, it is clear that only the WMNs are addressing the user level and infrastructural mesh connectivity, while the wired mesh networks are usually connecting infrastructural devices.

Today most of the user level devices are equipped with wireless technology, like Bluetooth [6-7] and Wi-Fi (Wireless Fidelity). For the infrastructural level the use of Wi-Max (Worldwide Interoperability for Microwave Access) [56] and Wi-Fi technologies are usually common, depending on the type, terrain, bandwidth requirements and interference considerations.

2.2 Wireless mesh networks

The study of WMNs is an active research field. WMNs are considered self-healing, self-optimizing and fault tolerant networks [42] [34] [23]. A WMN is a unique type of network, since it combines the characteristics of MANETs, WSNs and cellular technologies. Generally WMNs are considered to be a type of MANET. The similarities between WMNs and MANETs lie in the multi-hop nature and node mobility, but the differences include the following [26] [51]: the WMN nodes use gateways to reach the internet, wired LAN or other WMNs. Traffic flow in WMNs is usually between the client and the internet through the gateways, while in MANETs, the node can be the source and destination of the flow, although there are some cases of node to node communication in WMNs that are the same as MANETs [2]. WMN nodes can be stationary or mobile, but MANETs' nodes are usually expected to be mobile. WMNs' infrastructure and architecture are considered to be fixed or have very low mobility and are formed by access points, routers and gateways. WMNs do not consider mobile nodes as part of the WMN infrastructure, but MANETs' infrastructure is formed by mobile nodes, which are considered to be part of the WMN infrastructure. Backbone devices of WMNs are considered to be non-energy constrained devices, while MANETs' nodes are considered to be backbone and energy constrained devices.

WMNs make use of available natural resources, like air, to transmit the information in the form of modulated radio waves, without the use of wires, from one location to another. This makes it affordable for the least developed and developing countries [8]. The success of WMNs is that they allow computers to reach each other in public and rural areas. Directional-point-to-point antennas are also used to build backbone links. These make it possible to connect longer distances. Furthermore, vendors are very interested in producing equipment for this technology [44]. There is also wide spread industry support for wireless networks. Companies like Nokia, Microsoft, Motorola and Intel are actively working to support WMNs [51]. Reduction in wireless mesh equipment prices, as well as compatibility among them, is increasing, but there is lack of regulatory overhead on unlicensed frequencies. Major drawbacks for WMN are short signal range, interference (because of using unlicensed frequencies), the way the medium is used and that bandwidth is shared among the nodes. Wireless networks take advantage of natural resources and can therefore be deployed faster than wired networks. Low cost systems, based on wireless technologies, have been installed in rural villages, in South East Asia, resulting in a two times lower cost than that of wired solutions [21].

WMNs are implemented to address different types of user needs and provide various services. A study by [45] shows that hundreds of cities are planning and implementing WMNs for public Internet access, safety and businesses. Vendors must also focus on providing QoS, address issues related to mobility and support public and business applications. The author also states that some wireless mesh products are focusing on layer two, for providing better QoS, while some vendors focus on layer three's instant routing products, to provide QoS and achieve zero downtime. Today people are expecting to be connected anytime and anywhere with their mobile phones, PDAs, laptops and other hand-held devices. This is a great opportunity for businesses to introduce more wireless enabled devices in the areas of health, education and entertainment [42]. Development in the multi-media computing environment has resulted in producing networking devices which have the capability to carry different types of traffic. The types of traffic include text, images, audio and video. Carrying audio and video across wireless links, which is referred to as real-time traffic, requires QoS for prioritization [41]. For this reason the IEEE 802.11e MAC (Media Access Control) protocol was proposed to provide QoS on the MAC layer. In the MAC layer, it uses the CSMA/CA (Carrier Sense Multiple Access /Collision Avoidance) mechanism to avoid collision and to share the medium, based on single hop transmission. Research done by [46] and [39], also confirms that since WMNs are characterized as multi-hop transmission, the IEEE 802.11e which deals with QoS, does not fit the requirements of WMNs.

WMNs are defined by the IEEE 802.11's standard. This standard is based on 802.11a, 802.11b, 802.11g and 802.11n which carry the actual traffic. This standard requires Hybrid Wireless Mesh Protocol (HWMP) to be supported by default on the mesh nodes, but it also allows the use of other link state routing protocols, like OLSR (Optimized Link State Routing) and BATMAN (Better approach to mobile ad-hoc networking), and even static routes are supported. In this standard, the mesh nodes are called Mesh Stations (Mesh STAs). Mesh STAs are allowed to build links with their neighboring Mesh STAs or access points. Mesh STAs are considered to be power-constrained devices, able to relay traffic to other destined nodes. This standard and its amendments are expected to be applied to software only and no changes are required on the existing 802.11 chipsets and hardware. This standard is still in its preliminary stages, but there are some products, like the One Laptop per Child (OLPC) project, that have implemented the 802.11s draft [43].

Today wireless networks are extended by another alternative: multi-hop WMNs. Mesh networks are formed by stationary/mobile wireless clients or routers that can be from different vendors. Mesh network deployment is easy and requires less maintenance and administration and once more users join the network, more wireless routers can be added to result in an extended coverage area. The performance of the network could be affected by delay, jitter, packet loss, the number of hops, number of channels, clients, routers, antenna-positioning, mobility, throughput and application requirements [34].

Wireless networks can be divided into four main categories. They can be single-channel single-hop, multiple-channel single-hop, single-channel multiple-hop or multiple-channel multiple-hop [32]. From these four categories, only the latter two, which are single-channel multiple-hop and multiple-channel multiple-hop, are considered as the basis for the WMNs formation.

Single-channel single-hop wireless networks are formed by one channel and one hop. In this type of network, as the number of users increases, the overall throughput decreases. Research has been conducted by [47] in 802.11b standard, using one access point, nine laptops and five PCs. Traffic generation was done through customized software that generated UDP packets. This research shows that as the number of nodes increases, the overall throughput decreases. This research also shows that network performance is more dependent on the wireless card implementation than the nodes' processing power and capacity.

Multiple-channel single-hop wireless networks are formed by nodes communicating with each other directly, establishing a peer-to-peer connection among the nodes. Each node sends the data independently. Interference and collision among the channels are not possible, since each device is using a separate data channel. A study by [5] suggests that energy-efficient routing protocols can result in uniform energy usage by the nodes. These types of networks perform well, but building such a network is very expensive, since it requires non-overlapping data channels for each node's communication. The process will therefore be costly for the end users, despite the fact that it results in a single-hop approach which does not provide redundancy and multiple paths.

Single-channel multiple-hop is a single-channel multiple-hop wireless network. It is also called an ad-hoc or mesh network [22]. In this type of wireless network, the mesh nodes use one shared channel across the network. The medium bandwidth is divided among the nodes using the same channel. Bandwidth allocation to each node is lower compared to multiple-channel multiple-hop networks, but this type of network is easier and cheaper to build. If bandwidth-hungry applications are properly managed, this type of network performs well, but if the network is expanded greatly and the number of nodes increases, then the mesh node performance degrades and all nodes experience lower bandwidths, packet loss, higher latency and jitter.

Multiple-channel multiple-hop wireless networks are also called ad-hoc or mesh networks. They are good for capacity enhancement and can use multiple channels on multi-hop mesh networks, instead of using a single chunk of 20 MHz frequency that is provided by the LAN standards. By using multiple channels, the sending stations can make use of the two channels that are available and can send the data. On the receiving end, the station can receive and send, using multiple channels. Such a system can provide more capacity than the single channel provided that appropriate protocols are used [4] [11]. Research [44] shows that a solution to achieve higher bandwidth, less delay and jitter when using multiple channels in a multi-hop network, is to use separate channels for clients and for backhaul ingress and egress traffic, in order to achieve full duplex bandwidth. Another experimental research project by [32], which studied the performance of 802.11b wireless mesh multi-hop backbone networks, regarding multiple channel usage, shows that using non-overlapping channels on multiple radios, gives a better performance than the single-radio implementation. The round trip time (RTT) was lower in two-cards multiple channel than two-cards one channel and one-card one channel. The experiment also shows that for WMNs non-overlapping channels can provide better throughput, which is the total amount of data that is successfully transmitted to the user.

According to the above research, using multiple channels in multi-hop mesh networks, provide better bandwidth and better round trip time than single channel multi-hop networks. Unfortunately building a mesh network with multiple channels is expensive, because a separate radio is needed for each channel that the device is using. For example, if 5 wireless routers and 10 wireless clients are used in a WMN, and multiple non-overlapping channels of 1, 6 and 11 are used, three radio transceivers are required on each device:(5x3) +(10x3)=15+30=45. As a result it becomes very expensive to build such a mesh network, unless new radio technology will allow using multiple radio frequencies in a single radio.

Studies also show that multi-channel multi-hop networks are not good choices for VoIP applications that use a round robin schedule for switching between multiple channels, because the time needed to switch between channels, does not meet the VoIP requirements [10].

A better, more practical and cheaper approach to build WMNs, is to use a single-channel and multi-hop. This requires a single radio frequency for all devices participating in the mesh network. If QoS mechanism is properly implemented, the problem with delay, jitter, packet loss and better bandwidth can be addressed effectively. Since WMNs have a dynamic formation with multiple paths for the packets, and links are coming up and going down frequently and changing their positions by being mobile, the convergence and synchronization of the routing table requires a routing protocol that can adjust itself to the network topology and prevent routing loops in mesh networks. Mesh routing protocols' choice and characteristics can affect the QoS [26]. There are several important factors for choosing a mesh routing protocol, like the size of network, the nodes' mobility and type of traffic. Mesh routing protocols are usually classified as proactive, reactive or hybrid.

Mesh networks require resource management in order to provide better QoS. There are three areas that can be addressed by mesh resource management processes. Firstly, network configuration and deployment, secondly, routing, and lastly mobility management and admission control [51]. There are many mesh routing protocols, but some of them are more commonly used. These protocols are: BATMAN (Better approach to mobile ad-hoc networking), DSDV (Highly Dynamic Destination-Sequenced Distance Vector routing protocol), HSR (Hierarchical State Routing protocol), ZRP (Zone Routing Protocol), IARP (Intrazone Routing Protocol/pro-active part of the ZRP), WAR (Witness Aided Routing), OLSR (Optimized Link State Routing Protocol), DSR (Dynamic Source Routing), TORA (Temporally Ordered Routing Algorithm), CGSR (Clusterhead-Gateway Switch Routing), GeoCast (Geographic Addressing and Routing), DREAM (Distance Routing Effect Algorithm for Mobility), LAR (Location-Aided Routing), AODV (Ad-Hoc On Demand Distance Vector Routing), FSR (Fisheye State Routing), TBRPF (Topology Broadcast Based on Reverse Path Forwarding), LANMAR (Landmark Ad-Hoc Routing Protocol), GPSR (Greedy Perimeter Stateless Routing) [51]. Another routing protocol, GoDRP, is also used by simulation software, like ns2 (Network Simulator 2) and NCTUns, to calculate the routes based on nodes' position and signal range, without any routing protocol overhead. This type of routing protocol is used to benchmark the simulations to the best way a routing protocol can theoretically perform [15] [52].

2.3 VoIP QoS

There are many services provided by computer networks. One of the most used services is VoIP. Using a computer network for VoIP communication is relatively cheaper, since it uses existing infrastructure and the users can use the service from anywhere, using various devices like VoIP phones, PDAs (Personal Digital Assistants), laptops, desktops, smart mobile phones, etc. VoIP applications have unique characteristics, because of its real-time mode of communication, small packets, sensitivity to delay, jitter and packet loss [40] [25]. Although VoIP applications have been widely used for more than 20 years, there is still a lack of QoS for VoIP applications in the new emerging networks. VoIP applications are unique in that they exchange many small packets which are made of big packet headers and small VoIP payloads, but little effort has been made to address and investigate these problems on wireless multi-hop networks [28] [29].

VoIP requires call admission methods to admit the call. A study by [3] addresses two questions in WMNs to support VoIP. Firstly, in order to maintain QoS, the call admission has to be studied. This can be done by measurement-based models that can model the available capacity and can help in call admission decisions. The second question is to find a feasible route, considering the ratio of interference and carrier sensing ranges. This evaluates the path to see if it is feasible for the route selection processes. This study also suggests that new routing metrics, like max residual feasible path and routing, using call statistics, should be used in order to improve the VoIP performance.

VoIP traffic has its own characteristics regarding its packet size, number of packets per second, inter-packet delays and it is also dependent on the type of codec being used. VoIP characteristics, as explained by [13], show that on the G.729 VoIP, payload can be 10, 20, 30 or 40 bytes, but the default is 20 bytes. Since VoIP uses the RTP on the application which uses a header of 12 bytes, UDP on the transport layer with a header of 8 bytes, and IP on the network layer, with a header of 20 bytes, it totals 40 bytes of RTP/UDP/IP headers. Now, if a VoIP payload of 20 bytes is added, it totals to 60 bytes, without the consideration of data link headers. G.729 codec with the 20 bytes VoIP payload requires 50 packets to be sent per second. The number of packets can change if the VoIP payload increases or decreases, but for a normal calculation, 50 packets per second is considered to study the VoIP traffic. Also VoIP conversations have speech periods and silence periods. Studies show that if VoIP applications use the voice activity detection, and during the silent periods, voice packets are not sent, it saves the bandwidth about 35% for an average volume of 24 calls simultaneously. Studies [54] show that the VoIP inter-packet delay, which is delay time between transmitted packets, and the next packet in the queue, is usually between 10 and 30ms. The inter-packet delay time can even increase when the back-off algorithms sense that the medium is busy.

Researches have tried to identify VoIP flows in real time, in order to manage network traffic issues, prioritize VoIP flows, reserve bandwidth or block calls to certain destinations. These efforts face a number of challenges, like usage of non-standard protocols or ports, non-standard codecs, payload encryption and silence suppression. The research shows that the nature of human conversation makes the VoIP traffic patterns different and unique, compared to other applications. Therefore, a flow identification using the human conversation can provide more promising results than other VoIP flow identification methods. The study also shows that when two persons, A and B, talk to each other, their conversation can be modeled in four states: A talking, B talking, both A and B talking, both silent. The Markov 4-state chain can model these states.

Studies show that VoIP conversations are made up of periodic talk-spurts and silence gaps, since human conversations have talk periods and silence periods. A study conducted by [16] states that voice conversations can be distinguished by two types of voice streams, considering different voice codecs. One type of codec generates constant bit traffic streams, like the G.711 codec, while the second type of stream uses the codec that uses the silence suppression and generates active (on) and inactive (off) streams on the G.729 B and G.723.1 codecs. From a modeling point of view, the second type of stream has significance and most technologies use this type of streaming to translate a human conversation into a VoIP stream.

WMN QoS is usually affected by delay and packet loss [18]. Usually one-way delay of 200 ms, and less than 5% packet loss, is acceptable in VoIP conversations [3]. Besides, the IEEE 802.11e standard which addresses the QoS, is not designed for ad-hoc WMNs, therefore VoIP implementation in WMNs cannot be supported by QoS standards [39].

Researchers are suggesting ways to improve QoS. A study by [18] on 15 mesh nodes, using the common G.729 codec, with 20 byte VoIP payload, transporting 50 packets per second, shows that by taking the VoIP silence period into account, the utilization of the bandwidth can be increased by up to 30%. The silence periods, where no packets are sent, are natural in VoIP conversations. This study also shows that in order to improve service quality and mesh network capacity, several methods, like multiple interfaces, label-based forwarding architecture and packet aggregation, can be used. Among all the methods, this research emphasizes that label-based forwarding is the most appropriate method for improving the VoIP applications. Another study by [28] suggests that using packet aggregations in IEEE 802.11, WMNs can be a promising solution for VoIP applications. Since VoIP applications are using many small packets with huge headers, if multiple small packets can be assembled in one packet with an aggregated header, it will reduce the MAC layer and physical layer overheads, and it will save the transmission time.

2.4 Wireless mesh and VoIP QoS

WMNs are used by various real-time applications, including VoIP. Research show that the critical metrics and factors that affect QoS in WMNs are delay, jitter, packet loss and bandwidth [34] [40] [25]. There are other hidden factors as well, like mobility, obstacles and weather conditions that affect the link quality [30]. Although the 802.11T standard for measuring QoS is underway, there is still a lack of industry standards for measuring wireless mesh QoS factors [34]. Mobility is one of the main factors of measuring mesh QoS [1] but it is one of the most complicated and challenging factors to measure. Also, WMNs lack resource management, which results in poor QoS for end users. Therefore, QoS issues require innovations and research [51] [2] [25].

Prior to implementing VoIP applications, it is important to understand and test whether the existing networks can support VoIP applications. A study by [35] states that VoIP applications are susceptible to delay, jitter and packet loss. These factors can make the VoIP applications unacceptable for average users if they are not in the applications' acceptable ranges. Jitter with variation of less than 100ms, can be afforded by jitter buffers. The acceptable rate for packet loss varies from codec to codec, but the general trend should be to achieve zero packet loss.

Multi-channel multi-hop mesh networks are not feasible to support VoIP applications. A study by [10] shows that mesh nodes need to switch from one channel to another channel in order to communicate with the neighbors, but the current switching schedules do not consider the requirements of real time traffic, like VoIP. When the nodes switch from one channel to another channel, there are several phases before the new channel is used and each phase takes its own time. These phases include: sending buffered frames into the hardware queue of the Network Interface Card (NIC); stopping interrupt service routines and sensing the medium; the DCF (Distributed Coordination Function) protocol with RTS/CTS (Request to Send/ Clear to Send), which requires time, due to back-off algorithms, before data is sent to the newly switched channel. If the duration of all these phases and processes is added, it will not make the multi-channel multi-hop networks favorable for VoIP applications. The study suggests that instead of the round-robin method, a new QoS-aware scheduling method, which uses the packet header to prioritize traffic, can to be used.

Packet aggregation can even result in further delay. Research by [33] explains VoIP applications work step by step, by first sampling voice signals, digitizing and encoding them, then encapsulating the encoded data into packets, using the RTP/UDP and IP. The de-packetization process happens on the receiver’s end and the data is forwarded to the play-out buffer in order to compensate for the resulted jitter. There are various methods to boost the VoIP performance, but using the packet aggregation on the G.729 codec with 20 bytes of VoIP payload and 50 packets per second, can reduce the bandwidth utilization, due to large protocol headers. Although these protocol headers can be aggregated, it can result in further delay in some cases, because once the VoIP packets are generated, they are not immediately transmitted, but kept for other packets to aggregate with.

Nodes’ mobility can affect the performance of mesh routing protocols and QoS. A study by [1] explains that the movement of the ad-hoc devices, which is referred to as nodes’ mobility, can introduce a number of challenges, like network topology changes, increase in frequency of route disconnections and packet loss that affect QoS. The study suggests a Speed Aware Routing Protocol (SARP) to be used. This protocol reduces the effects of high mobility, and results in a reduced number of route disconnections and packet losses.

Before a mesh routing protocol is selected, the node mobility model has to be identified. An empirical study by [17] compared DSDV with DSR mesh routing protocols. The comparison of these routing protocols was done on four-node mobility models. These are Random Waypoint, Random Point Group Mobility, Freeway Mobility and the Manhattan Mobility models. The Manhattan Mobility model, explains the mobile nodes’ movement pattern in urban areas with vertical and horizontal streets. In this research, UDP traffic was generated and exchanged among the mesh nodes. The research results show that DSR performs better than DSDV in high mobility networks, since DSR has faster route discovery, compared to DSDV, when the old route is not available.

Researchers are actively working to find better solutions for the provision of QoS in WMNs, which is a challenging task [40] [25]. In wireless networks, when the mobile nodes move around the cells or to another access point, the session that is currently in progress, should not end or drop off while the node is trying to receive the necessary service from the new access point. The QoS provisioning should be checked prior to allowing the node to be associated with new access point. This process is controlled by the call admission control (CAC) algorithm. This algorithm assigns bandwidth to new connections, based on the traffic type of the new connection and the current connections. The challenge to the algorithm is that it should admit as many connections as possible, considering the minimum call dropping rate, low association, handover latency and efficient bandwidth allocation. Another method to provide QoS is to adapt scaling of bandwidth rate, based on the traffic priority [1] [7], but this method requires significant computation on the heavily loaded networks, which results a considerable delay [20].

Statistical reference models can be created off-line to help select better routes for the mesh nodes. The proposed solution by [12] is the conservative and adaptive quality of service (CAQoS) method. This method's central focus is a statistical reference module (SRM), which provides QoS. The SRM decides how and when the bandwidth should be scaled-down for the new connection. It gathers information from the application profile, type of traffic and the status of the network, from both the neighboring base-station and the mobile terminal in three intervals (peak, moderate and off-peak). It then creates three separate QoS provision models, according to the number of calls and their traffic conditions over a period of time. These models are then referenced in future call admissions. Each model is created and optimized off-line. Therefore, while calls are in progress, they are not affected by the delays at the time of creating these models.

In order to measure the QoS factors, different methods and tools are used to conduct the tests. Some researchers have developed their own testing and measurement tools, while some researchers have used on-the-shelf tools, like Iperf, Rude and Crude, STG (Send Traffic Grapher), and RTG (Receive Traffic Grapher) [52].

Measuring delay can be done by either calculating one-way or two-way delay. One-way delay is the time that expires between a packet entering the mesh backbone and leaving the mesh backbone. One-way delay can also be used as routing metric to select the optimal route for the real-time traffic. There are two simple methods of measuring one-way delay. One method is to add a time stamp to each packet that is sent and subtract the reception from the transmission time, or to use the Round Trip Time (RTT) and divide it by two, to find the one-way delay. The second method is to send pair packets (PP) one after the other, and timestamp each packet. Packets are queued, and the time between when the first packet is received and the second packet is processed and calculated. The problem with the above two methods is that when the time stamp is used, all the nodes' clocks must be synchronized, either by NTP (Network Time Protocol) or GPS (Global Positioning Satellite), which is practical in a test bed, but not in 'real' life. Furthermore, the measurement results with RTT do not show each forwarding node's delay. The other problem is that probe messages are smaller in size and do not know how to simulate the real payload. Another algorithm is the adaptive per hop differentiation (APHD). This algorithm uses the packet headers to calculate the inter-node (time it takes for a packet to move from one node to the next node) and the intra-node (the processing time of a packet in a network card driver of a node, till it gets routed to the next node). APHD uses the packet header fields to calculate the delay and hops up to this point, while the packets get routed within the mesh network [25].

Measuring Jitter can be done in various ways. Jitter can occur between the wireless mesh nodes in a bit level or packet level. One reason is that the circuit that is being used for voice and video is shared among other applications as well. So, on the same channel or medium that the voice or video traffic is transported, other data is also transported. This can result in resources being used by other traffic, while the voice and video traffic will have to wait their turn in the queue [27]. The process results in the introduction of jitter among the packets, leaving the queues at different time intervals and reaching their destinations at various times. A study by [48] explains acceptable jitter rate for voice is less than 100ms. If the rate goes beyond this limit, it becomes difficult for the jitter buffer to compensate and VoIP quality starts degrading. Jitter can also be calculated by using timestamps on the IP header; the time that a subsequent packet reaches its destination is subtracted from the time that a previous packet reaches it, and jitter between successive packets can be estimated.

Packet loss measurement can be done in several ways. Packet loss can occur due to various reasons. It can be due an unreliable medium, unreliable protocol, buffer overflow, resource limitation, link congestions, channel errors, contention between hops [19], etc. A packet loss can result in portions of data not reaching the destination, and since most of the real-time applications use the UDP as the transport protocol, data recovery will not happen if packets are lost. As a result, the voice or video quality will degrade. Packet loss can be calculated by comparing the number of sent packets from the source node with the number of received packets at the destination node.

Measuring bandwidth can be done with monitoring software that shows the device's interface usage. Usually Simple Network Management Protocol (SNMP) is used to monitor the interface usage. The unit of measurement for the bandwidth is bits per second (bps). Since links are usually working with bandwidths above 1024 bps, measurements are usually done using Kbps (Kilo bits per second), Mbps (Mega bits per second) and Gbps (Giga bits per second). The achievable bandwidth is called throughput. Throughput is always less than, or equal to the maximum bandwidth of a link. The best practice to calculate bandwidth is to transfer a large file over a link and divide the file size by the amount of time elapsed to download the file, which will show the real data rate of a link. This is referred to as the goodput. Goodput is usually less than bandwidth and throughput.

2.5 Summary

In this chapter the research community's efforts on VoIP implementation over WMNs with an emphasis on QoS, was explored. The literature shows that WMNs' multi-hop nature, medium usage method and the lack of QoS mechanism in the mesh nodes, lead to introducing more delay, jitter and packet loss. Each mesh node treats a VoIP and a non-VoIP packet equally and priority is not given to the sensitive traffic. Although the research community is actively working on ways to solve these problems, their proposed methods are not yet fully addressing all these issues. Therefore, VoIP applications are still facing a number of challenges in WMNs. These challenges are identified as increased delay, jitter, packet loss and throughput in WMNs. Therefore more research on the subject is required to determine which of VoIP QoS factors are most affected and how these factors impact the VoIP quality, considering the WMNs' implementation scenarios. In order to investigate this, the literature survey discussed WMNs' characteristics, WMNs' scenarios and VoIP's characteristics. Based on the characteristics of WMNs and VoIP, traffic profiles can be designed in order to simulate and test the resulting impacts on QoS. The next chapter focuses on the research methodology, problem statement, VoIP and non-VoIP profiles, simulation software and simulation cases.

3 Methods

The research framework used to analyze VoIP QoS in WMNs is presented in this chapter. This chapter's structure is as follows: Section 3.1 presents the research approach and the problem statement. Section 3.2 presents the experimental design, the simulation environment, mesh topology, mesh routing protocol, VoIP and non-VoIP traffic profiles. Section 3.3 presents the WMN simulation scenario designs, considering three scenarios. These three scenarios are the no mobility, limited mobility and full mobility scenarios. Section 3.4 presents the data collection methods and techniques. Finally, Section 3.5 summarizes this chapter.

3.1 Research approach

Referring back to related work, it was stated that the IEEE 802.11e standard, which addresses QoS in wireless networks, is designed for single-hop. Since WMNs are of a multi-hop nature, the IEEE 802.11e standard cannot be applied to them, because it can lead to more delay, jitter, packet loss and improper bandwidth allocation issues, compared to single-hop [44],[40],[25]. It was shown that these factors cause QoS problems for VoIP applications. WMNs' unique characteristics, due its dynamic formation, multi-hop nature and node mobility, were discussed in the previous chapter. VoIP traffic's unique characteristics, which are due to human conversation nature and the way conversation is translated, using various codecs and the presence of talk periods and silence periods, were also shown. Implementing VoIP applications in WMNs introduces a number of challenges and issues with QoS. Therefore, this research focuses on the following question: **How is VoIP applications' QoS affected by wireless mesh node mobility?**

The methodology for research in Computer Science is usually defined as either theoretical or empirical (experimental). Theoretical research is usually conducted by collecting mathematical, logical and conceptual proof. Empirical research is conducted by the "building of, or experimenting with or on, nontrivial hardware or software systems" [36]. Since the aim of this research is to discover the problems that VoIP applications experience in WMNs, and to measure quantities of delay, jitter, packet loss and bandwidth, an empirical study is required in order to discover the actual effects of these factors on service quality in WMNs [37]. Figure 3-1 summarizes the over view of research approach. This methodology is based on an experimental study, as used by [47] [32] [17]. In this research, wireless mesh nodes are used to produce VoIP traffic, using a single-channel multi-hop mesh network. Data collection, measurements and statistics are done on source and destination nodes. The focus is on finding the amount of delay that is caused by the number of hops; amount of jitter produced as a result of multiple hops and transmission delay; the number of packets lost among the nodes; and bandwidth used at each node by VoIP applications.

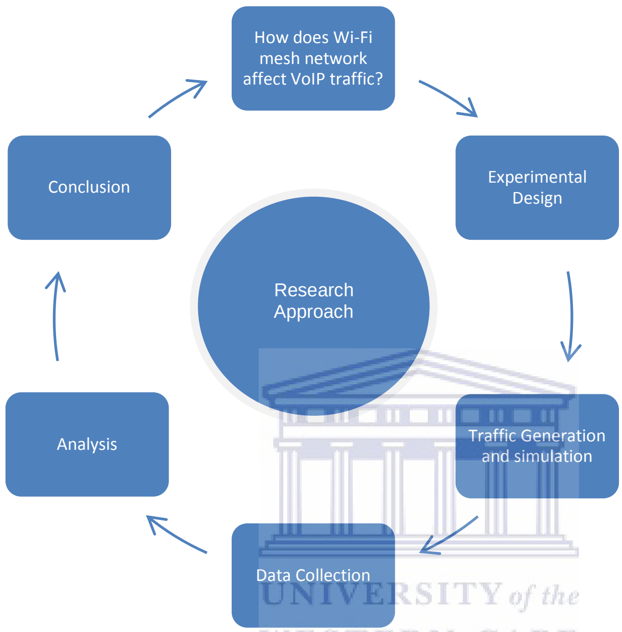


Figure 3-1 *Overview of research approach*

The literature survey in Chapter 2 showed that not much work has been done by other researchers addressing these types of questions and problems in WMNs, especially when the mesh nodes are partially or fully mobile. The literature survey also indicates that QoS for VoIP traffic still remains a problem among the research community and more research is required to understand and investigate how the WMN affects different types of traffic. This research's aim is to investigate and discover the problems that can affect the VoIP implementations. VoIP implementation will be studied in three different WMN scenarios. Since the 802.11e is not supporting QoS in WMNs, there will be no QoS provisioning done among the wireless mesh nodes. This thesis analyzes the major factors which may result in poor QoS in real-time applications, like delay, jitter, packet loss and bandwidth, if they are not within acceptable limits. The analysis will be based on these major QoS factors for VoIP traffic only, and this research aims to discover whether VoIP applications can be successful in various scenarios in WMNs.

3.2 Experimental design

The experimental design in this research considers the usage of a single-channel multi-hop WMN on 802.11b IEEE standard. This standard has been selected on the basis of its wider availability on various devices. By using this standard in the experimental design, it is easy to discover how the lower bandwidth rates affect the VoIP traffic in WMNs, because higher bandwidth standards usually allocate more bandwidth to transport more data among the nodes, but lower bandwidth standards have to work with complex queues, and deal with delay, jitter and packet loss. The 802.11b standard will be deployed among 26 nodes, which will be simulated by simulation software, named NCTUns version 6.0. Going forward, Section 3.2.1 presents the simulation environment, Section 3.2.2 discusses the mesh topology, Section 3.2.3 explains which mesh routing protocol is used in this research and Section 3.2.4 presents the VoIP and non-VoIP traffic profiles.

3.2.1 Simulation environment

In order to build a WMN for conducting various experiments, the NCTUns simulation/ emulation software [53] was selected, because the simulation software has the capability to simulate WMN and it has a better interface and is less complex compared to other simulation tools. With this simulation software WMNs have been designed and test cases were set up in order to analyze VoIP traffic behavior on WMNs. The NCTUns 6.0 simulation/emulation software simulates the wireless mesh clients. It also facilitates the use of real-world traffic generation and monitoring tools. Nodes' mobility is simulated with the nodes moving at an average walking speed of 1.3m per second [31]. In this research wireless mesh nodes are considered as part of the mesh backbone, like in MANETs. The IEEE 802.11s standard for WMNs considers the mesh nodes as part of the network infrastructure. Mesh nodes will be stationary or mobile and the WMN formation will only be made up of wireless mesh clients. Wireless access points and wireless routers are not considered to be part of the mesh setup in this research. The mesh network is more of an ad-hoc type of network. The test cases have been designed in three different scenarios. Each scenario is tested against two VoIP traffic profiles. Profile one is a simple VoIP conversation between two mesh nodes without any background traffic. Profile two is VoIP traffic along with background traffic simulated by Transmission Control Protocol (TCP) in greedy mode. These scenarios are further explained in the experimental design.

The traffic generation tools are used to generate traffic and monitor each node's behaviour, considering the QoS factors. The NCTUns 6.0 only runs in Fedora 12, version i386 and with kernel version 2.6.28.9. Once the NCTUns 6.0 is installed, it will create its own NCTUns kernel. After the NCTUns installation is completed, the user has to reboot the machine. Once the machine reboots, the user can choose to load the NCTUns kernel, as shown in Figure 3-2.

Figure 3-2 *Selecting NCTUns kernel*

The Fedora 12 will use the NCTUns kernel and this will enable the NCTUns to work on its own kernel and enable the simulator to load its required modules. The NCTUns 6.0 has three components that run together to make the simulator work. First, the dispatcher module, shown in Figure 3-3, and then the coordinator module, shown in Figure 3-4, must run from the root user. Once the dispatcher and the coordinator run into the memory, then the NCTUns client module, shown in Figure 3-5, must run from the normal user. Once these three components are loaded successfully, then the NCTUns interface can be used, as shown in Figure 3-6.

The NCTUns is a network simulator and emulator, capable of simulating various protocols used in both wired and wireless IP networks. Its core technology is based on the novel kernel re-entering methodology invented by Prof. S.Y. Wang, while pursuing his Ph.D. degree at Harvard University (URL: http://nsl10.csie.nctu.edu.tw/)

```
[root@MTM bin]# ./dispatcher
ServerSocket listen to port:9810
ServerSocket listen to port:9800
(Active:0| fd:3)    (Active:1| fd:4)
--->New Server
Server accepted: 5       2
(Active:0| fd:3)    (Active:1| fd:4)    (Active:2| fd:5)
[From Server...] register|127.0.0.1|9830|9840|IDLE
(Register Complete!)
<ServerList>:127.0.0.1:FREE
(Active:0| fd:3)    (Active:1| fd:4)    (Active:2| fd:5)
```

Figure 3-3 *NCTUns dispatcher*

```
[root@MTM bin]# ./coordinator
/usr/local/nctuns/bin/
ServerSocket listen to port:9830        FD:4
ServerSocket listen to port:9840        FD:5
ServerSocket listen to port:9880        FD:6
UnixDomainSocket Bind Path:/tmp/nctuns  FD:7
[To Dispatcher...]      register|127.0.0.1|9830|9840|IDLE
[From Dispatcher...]    OK
```

Figure 3-4 *NCTUns coordinator*

```
[tariqmeeran@MTM bin]$ ./nctunsclient
mkdir /home/tariqmeeran/.nctuns
mkdir /home/tariqmeeran/.nctuns/etc
mkdir /home/tariqmeeran/.nctuns/tmp
Openfile:/usr/local/nctuns/bin/mesh6.sim/mesh6.tcl----
Openfile:/usr/local/nctuns/bin/mesh6.sim/mesh6.tcl----
Str2Mrt() is called, in_filename /home/tariqmeeran/.nctuns/mesh6mrt.tmp out_filenam
e /usr/local/nctuns/bin/mesh6.sim//mesh6.mrt single_filename /home/tariqmeeran/.nct
uns/sh_connectivity.tmp multi_filename /home/tariqmeeran/.nctuns/mh_connectivity.tm
```

Figure 3-5 *NCTUns client*

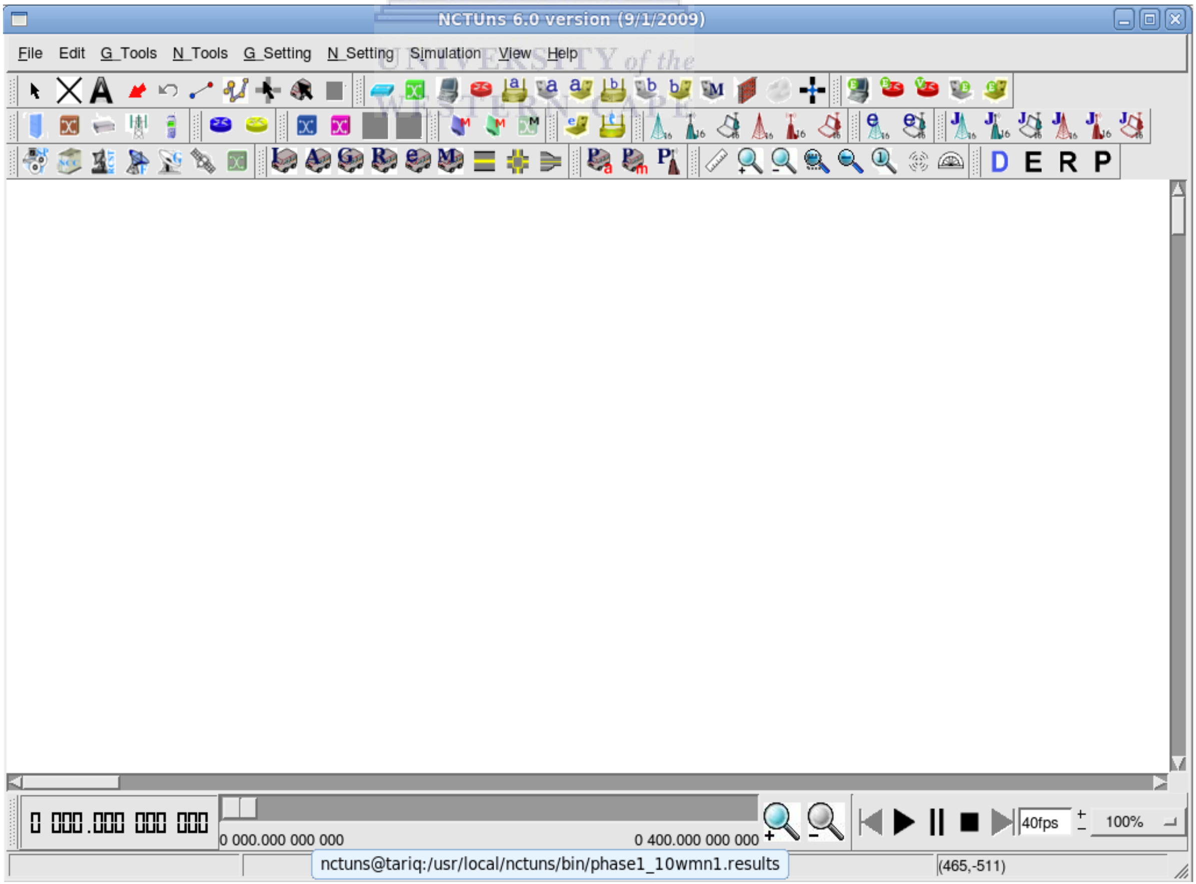


Figure 3-6 *NCTUns 6.0 workspace*

3.2.2 Mesh topology

In this research, WMN scenarios consist of 26 nodes, as shown in Figure 3-7. All the nodes are working in the ad-hoc mode and paired randomly and manually. The 26 nodes cover an area of almost 132248 m^2 (*y = 488m*, *x= 271m*).

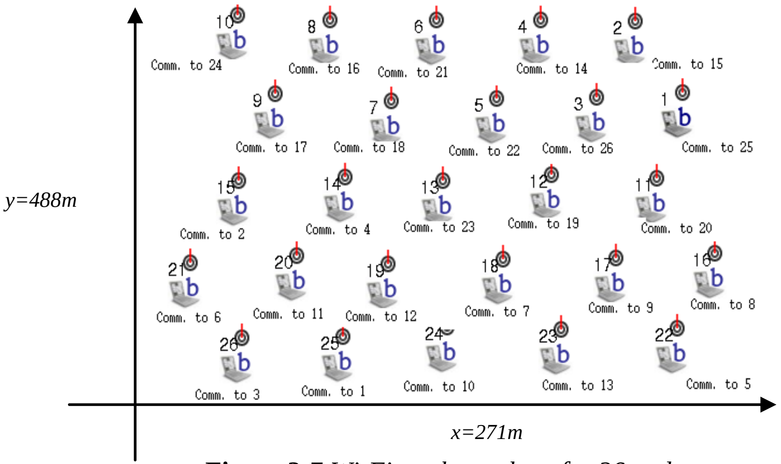


Figure 3-7 *Wi-Fi mesh topology for 26 nodes*

Each node's physical and channel model parameters are set according to Figure 3-8.

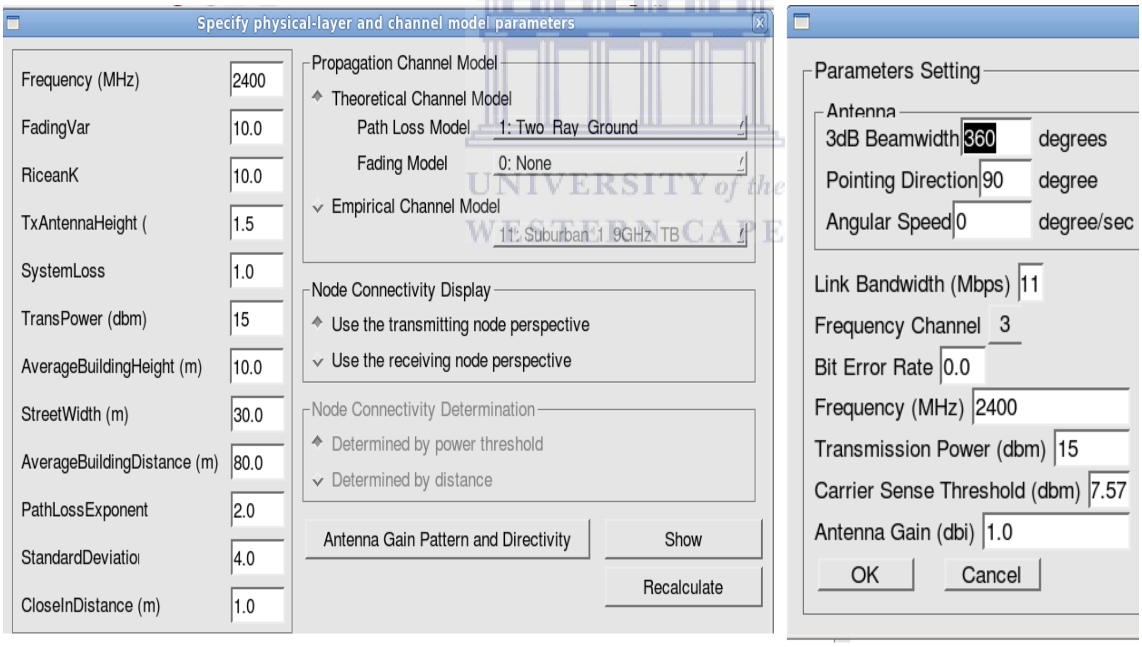


Figure 3-8 *Mesh node's physical layer and channel model parameters*

3.2.3 Mesh routing protocol

The NCTUns simulation software is running the God routing protocol (GodRP) in the mesh nodes. It calculates the best theoretically possible route to other nodes [15] [52], as discussed in the literature survey. In NCTUns the GOD's routing table is built using the single-hop and multi-hop routing tables, which is the most accurate method. GOD also calculates routes without introducing routing overhead(s). As seen in Table 1, the mesh nodes are communicating with each other randomly. Each mesh nodes is configured with an IP address: node 1's IP address is 1.0.1.1, node 2's IP address is 1.0.1.2 and node 26's IP address is 1.0.1.26.

Table 1 *VoIP random pairing*

Speaker 1	Node 1	Node 2	Node 3	Node 4	Node 5	Node 6	Node 7	Node 8	Node 9	Node 10	Node 11	Node 12	Node 13
Speaker 2	Node 25	Node 15	Node 26	Node 14	Node 22	Node 21	Node 18	Node 16	Node 17	Node 24	Node 20	Node 19	Node 23

3.2.4 Traffic profiles

A VoIP profile was designed to simulate two persons talking to each other, using a wireless mesh enabled device, such as a desktop, laptop, VoIP phone, handheld device, etc. For example, a mother talking to her child, where the mother does most of the talking, is simulated. During the VoIP conversation there are occasions when mother (speaker 1) and child (speaker 2) are both talking at the same time, mother is talking and child listening, child is talking and mother listening, or both mother and child are silent. While speaker 1 is talking, the mesh node is sending VoIP traffic to speaker 2 and the traffic goes across the mesh network to reach speaker 2. Mostly, when speaker 1 speaks, speaker 2 listens, and vice versa. Speaker 1 is selected in a sequences from 1 to 13 while speaker 2 is randomly selected. This model of communication is based on the Markov model, discussed in the literature survey.

As discussed in the literature survey, in a normal VoIP conversation, using the common G.729 codec, the voice coder sends 50 VoIP packets every second while the speaker is talking, but when the speaker is silent, no packet is sent [13] [3] [18] [29]. The VoIP payload can vary according to the codec setting, but by default the G.729 sends a payload of 20 bytes (20ms of VoIP conversation) in each IP packet. The payload size can vary between 10, 20, 30, 40 and 50 bytes. This research considers VoIP payload size of 20, 30 and 40 bytes. Therefore, in 10 seconds of a VoIP conversation, 500 packets must be sent. The literature survey shows that when the packets are being transmitted from the source to the destination, there is an inter packet delay (IPD) time. The inter packet delay time is considered to be between 0.01 and 0.05 seconds. VoIP software uses the RTP in order to transport voice traffic over UDP and IP. Each one of these protocols has its own headers. RTP has a header of 12 bytes, UDP has a header of 8 bytes and IP, a header of 20 bytes. If the size of all these headers is calculated, it will be 40 bytes in total. As the literature survey shows, a VoIP payload can be either 20, 30 or 40 bytes. If the RTP/UDP/IP headers are added to this, it will total 60, 70 and 80 bytes respectively. According to Table **2**, the packet size of 60, 70 and 80 bytes, will be considered in simulating the VoIP payload along with the RTP/UDP/IP headers.

The literature survey also shows that VoIP conversations are made up of periods when the speaker talks, pauses and listens. When the speaker talks, voice is detected by the codec and then transformed to digital form. Voice traffic is packetized and sent to the other party (listener). When the speaker pauses or listens to the other speaker, the voice detection algorithms used by the codec, recognize the pauses and listening periods and mark them as silent periods. During this time, no traffic is sent to the other party. The non-silence and silence periods are simulated as the ON and OFF modes where, in the ON mode, the packets are sent, and in the OFF mode, no packets are sent. This simulation is based on studies discussed in the literature survey.

In VoIP profile, the mother and child conversation starts with a short greeting. During the greeting, mother and child talk for almost 10 seconds each. Here both nodes are talking and generating traffic, therefore both are set to the ON mode. Then they pause for 2 seconds, the OFF mode, and then the mother starts talking for 30 seconds, the ON mode, while the child is listening, the OFF mode. This conversation continues for some time with a sequence of ON and OFF states. Then the mother says good-bye to her child, the child responds with a goodbye and the conversation ends. This conversation lasts 562 seconds. In total the mother generates approximately 18250 packets and the child generates 7500 packets for the whole conversation. These numbers are just estimated, and in real life applications, this number can change. When voice traffic is packetized, it can have varying sizes. The number may increase or decrease depending on the codec being used, considering the G.729 codec. VoIP profile for a human conversation is designed, using the Markov model as show in **Table 2**.

Table 2 *VoIP profile of human conversation*

Node A Mother (Speaker 1)										Node B Child (Speaker 2)								
Transmission status	Time in sec	Number of Packets	Time delay in sec between each packet		Packet size			Description	Traffic Direction	Transmission status	Time in sec	Number of Packets	Time delay in sec between each packet		Packet size			Description
			Min	Max	Avg	Min	Max						Min	Max	Avg	Min	Max	
ON	10	500	0.01	0.05	70	60	80	Mother Greeting	<-------->	ON	10	500	0.01	0.05	70	60	80	Child Greeting
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	30	1500	0.01	0.05	70	60	80	Mother talking	-------->	OFF	30	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	15	0	0	0	0	0	0	Mother listening	<---------	ON	15	750	0.01	0.05	70	60	80	Child Talking
OFF	2	0	0	0	0	0	0	Mother listening	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	45	2250	0.01	0.05	70	60	80	Mother talking	-------->	OFF	45	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	15	0	0	0	0	0	0	Mother listening	<---------	ON	15	750	0.01	0.05	70	60	80	Child Talking
OFF	2	0	0	0	0	0	0	Mother listening	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	60	3000	0.01	0.05	70	60	80	Mother talking	-------->	OFF	60	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	30	0	0	0	0	0	0	Mother listening	<---------	ON	30	1500	0.01	0.05	70	60	80	Child Talking
OFF	2	0	0	0	0	0	0	Mother listening	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	45	2250	0.01	0.05	70	60	80	Mother talking	-------->	OFF	45	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	15	0	0	0	0	0	0	Mother listening	<---------	ON	15	750	0.01	0.05	70	60	80	Child Talking
OFF	2	0	0	0	0	0	0	Mother listening	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	30	1500	0.01	0.05	70	60	80	Mother talking	-------->	OFF	30	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	15	0	0	0	0	0	0	Mother listening	<---------	ON	15	750	0.01	0.05	70	60	80	Child Talking
OFF	2	0	0	0	0	0	0	Mother listening	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	45	2250	0.01	0.05	70	60	80	Mother talking	-------->	OFF	45	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	15	0	0	0	0	0	0	Mother listening	<---------	ON	15	750	0.01	0.05	70	60	80	Child Talking
OFF	2	0	0	0	0	0	0	Mother listening	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	60	3000	0.01	0.05	70	60	80	Mother talking	-------->	OFF	60	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	30	0	0	0	0	0	0	Mother listening	<---------	ON	30	1500	0.01	0.05	70	60	80	Child Talking
OFF	3	0	0	0	0	0	0	Mother listening	----------	OFF	3	0	0	0	0	0	0	Child pauses
ON	45	2250	0.01	0.05	70	60	80	Mother talking	-------->	OFF	45	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother pauses	----------	OFF	2	0	0	0	0	0	0	Child listening
OFF	15	0	0	0	0	0	0	Mother listening	<---------	ON	15	750	0.01	0.05	70	60	80	Child Talking
OFF	2	0	0	0	0	0	0	Mother listening	----------	OFF	2	0	0	0	0	0	0	Child pauses
ON	5	250	0.01	0.05	70	60	80	Mother Good bye	-------->	OFF	5	0	0	0	0	0	0	Child listening
OFF	2	0	0	0	0	0	0	Mother listening	<---------	ON	2	100	0.01	0.05	70	60	80	Child Good bye
OFF	2	0	0	0	0	0	0	Mother Hangs up	----------	OFF	2	0	0	0	0	0	0	Child Hangs up
Total	564	18250								Total	564	7500						

In order to simulate a human conversation, a traffic generation tool that can simulate a human VoIP conversation by generating packets with varying sizes, varying inter packet delays and simulating ON and OFF periods, has to be used. To achieve this, STG and RTG tools have been selected. The STG tool is used to send traffic and the RTG is used to receive traffic. The STG can be used in several modes, like TCP, UDP and configuration. In this research STG is used with the configuration mode. In the configuration mode, a script can be written which can translate the human conversation in Table **2** into a form where the STG tool can read the script and generate the traffic as per the defined VoIP parameters for a human conversation. There are two STG scripts created for speaker 1 (Figure 3-9) and speaker 2 (Figure 3-10).

```
type: udp
start_time: 1
on-off: 1

on: packet: 500 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 2
on: packet: 1500 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 19
on: packet: 2250 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 19
on: packet: 3000 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 34
on: packet: 2250 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 19
on: packet: 1500 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 19
on: packet: 2250 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 19
on: packet: 3000 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 35
on: packet: 2250 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 19
on: packet: 250 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 4
end
```

Figure 3-9. *Speaker 1 STG configuration script*

```
type: udp
start_time: 1
on-off: 1

on: packet:  500 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 34
on: packet: 750 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 49
on: packet: 750 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 64
on: packet: 1500 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 49
on: packet: 750 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 34
on: packet: 750 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 49
on: packet: 750 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 64
on: packet: 1500 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 50
on: packet: 750 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 7
on: packet: 100 uniform 0.01 0.05 length: exponential 70 60 80
off: time: 2
end
```


Figure 3-10. *Speaker 2 STG configuration script*

The configuration file parameters for speaker 1's configuration script are explained below:

Type: udp — *It specifies which protocol to use. Since UDP is used, it is set to UDP type.*

start_time: 1 — *This sets the time when the stg should start transmitting the packets, defined as 1 second in this case.*

on-off: 1 — *This defines how many times this script should run. It is set to 1, since the script will run once from "on" till the "end".*

on: packet: 500 — *This sets the transmission mode to on and 500 packets will be sent.*

uniform: 0.01 0.05 — *This sets the inter packet delay time for a minimum of 0.01 seconds and a maximum of 0.05 seconds.*

length: exponential — *This sets the size of the packet, which is an average of 70, with a minimum* *of 60 and maximum of 80 bytes.*

off: time: 2 — *This sets the transmission to OFF mode, where no packets will be sent for a period of 2 seconds.*

A **non-VoIP profile** is simulated using the STCP (Sent Transmission Control Protocol) and RTCP (Receive Transmission Control Protocol) traffic generation tools. Here a simple TCP greedy traffic mode was used, where the tool establishes numerous TCP connections between the two communicating nodes and transmits TCP data and it is not limited to a file size. This traffic was transferred on the opposite flow to simulate the background traffic and keep all the nodes busy.

3.3 Scenarios

The test cases of VoIP only and VoIP, with non-VoIP traffic profiles, will be simulated in three different scenarios. These scenarios are organized in a series of sections. Section 3.3.1 presents the **no mobility scenario,** which is designed and configured to simulate all nodes in stationary/fixed mode. Section 3.3.2 presents the **limited mobility scenario,** which is designed and configured to simulate 10 nodes moving at a walking speed of 1.3m/sec, while the other 16 nodes are stationary. Section 3.3.3 presents the **full mobility scenario,** designed and configured to simulate all nodes moving at a walking speed of 1.3m/sec. Each scenario is tested against two types of traffic profiles and run 20 times to achieve accurate results. The first profile is scripted to generate VoIP traffic only and the second profile is scripted to generate VoIP and non-VoIP traffic. In profile 1, only RTP/UDP/IP traffic is generated by the simulation tool. In profile 2 both RTP/UDP/IP and TCP traffic are generated by the simulation tools. Figure 3-11 summarizes the scenarios and traffic profiles.

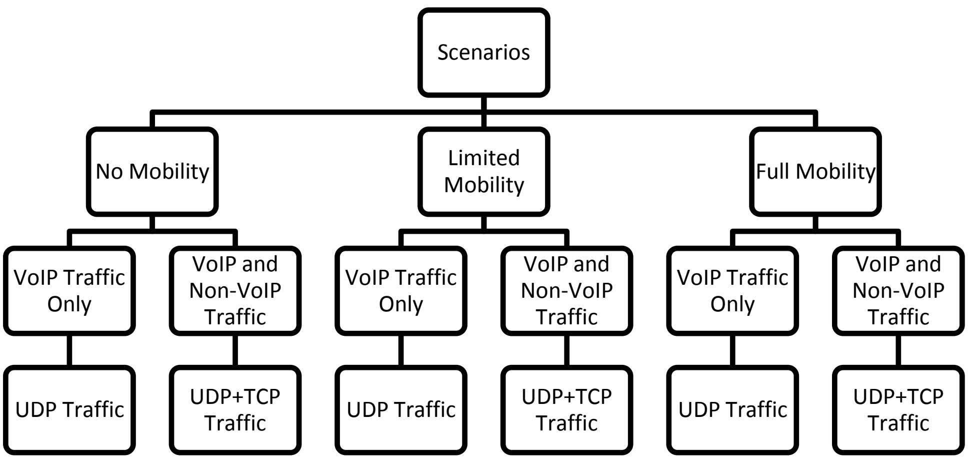


Figure 3-11 *Summary of different scenarios*

3.3.1 No mobility scenario

This scenario is designed to simulate a WMN with 26 mesh nodes. These nodes are a distance apart as seen in Table 3. The 26 nodes are all stationary and each node communicates with another node in the mesh network, i.e. there are 13 mesh node peers communicating with each other. Two traffic profiles on this type of WMN will be tested.

Table 3 *Mesh nodes' distances in stationary position in metres*

Node Name	N 1	N 2	N 3	N 4	N 5	N 6	N 7	N 8	N 9	N 10	N 11	N 12	N 13	N 14	N 15	N 16	N 17	N 18	N 19	N 20	N 21	N 22	N 23	N 24	N 25	N 26
N 1	**0**	74	80	145	173	236	271	333	379	419	75	140	234	321	419	138	152	217	309	384	479	198	228	294	376	458
N 2	74	**0**	73	88	144	185	236	284	340	370	133	150	223	300	292	208	199	234	307	371	461	260	268	310	379	452
N 3	80	73	**0**	83	93	162	191	256	299	341	88	77	159	243	340	171	136	161	239	309	402	207	197	237	309	386
N 4	145	88	83	**0**	77	97	154	196	254	282	171	130	162	224	311	254	211	204	249	301	384	288	260	272	323	385
N 5	173	144	93	77	**0**	86	98	168	206	250	163	81	85	155	249	241	174	135	171	228	316	255	203	197	245	310
N 6	236	185	162	97	86	**0**	79	99	162	185	244	168	135	157	228	324	260	210	209	236	306	342	284	258	279	322
N 7	271	236	191	154	98	79	**0**	88	108	160	255	161	82	77	157	327	247	168	137	156	231	327	248	196	203	242
N 8	333	284	256	196	168	99	88	**0**	80	86	331	243	171	132	159	408	332	257	211	200	241	414	337	280	266	279
N 9	379	340	299	254	206	162	108	80	**0**	74	262	265	172	95	80	430	345	253	177	137	162	422	331	253	213	207
N 10	419	370	341	282	250	185	160	86	74	**0**	413	321	236	168	139	468	406	320	251	209	212	486	400	326	287	271
N 11	75	133	88	171	168	244	255	331	362	413	**0**	98	199	290	389	82	77	158	260	341	439	127	155	231	320	408
N 12	140	150	77	130	81	168	161	243	265	321	98	**0**	101	192	291	165	92	84	169	246	344	174	130	160	236	318
N 13	234	223	159	162	85	135	82	171	172	236	199	101	**0**	91	190	260	173	86	86	149	244	250	166	123	160	228
N 14	321	300	243	224	155	157	77	132	95	168	290	192	91	**0**	99	349	261	162	83	79	160	334	238	157	134	165
N 15	419	392	340	311	249	228	157	159	80	139	389	291	190	99	**0**	447	356	254	155	82	82	426	325	229	163	132
N 16	138	208	171	254	241	324	327	408	430	486	82	165	260	349	447	**0**	92	197	304	389	488	72	156	256	353	445
N 17	152	199	136	211	174	260	247	332	345	406	77	92	173	261	356	92	**0**	105	212	297	396	82	78	167	262	354
N 18	217	234	161	204	135	210	168	257	253	320	158	84	86	162	254	197	105	**0**	107	192	291	172	79	77	162	251
N 19	309	307	239	249	171	209	137	211	177	251	260	169	86	83	155	304	212	107	**0**	85	184	274	170	76	74	149
N 20	384	371	309	301	228	236	156	200	137	209	341	246	149	79	82	389	297	192	85	**0**	99	359	253	152	80	85
N 21	479	461	402	384	316	306	231	241	162	212	439	344	244	160	82	488	396	291	184	99	**0**	456	349	245	154	79
N 22	198	260	207	288	255	342	327	414	422	486	127	174	250	334	426	72	82	172	274	359	456	**0**	108	214	311	405
N 23	228	268	197	260	203	284	248	337	331	400	155	130	166	238	325	156	78	79	170	253	349	108	**0**	106	203	297
N 24	294	310	237	272	197	258	196	280	253	326	231	160	123	157	229	256	167	77	76	152	245	214	106	**0**	97	191
N 25	376	379	309	323	245	279	203	266	213	287	320	236	160	134	163	353	262	162	74	80	154	311	203	97	**0**	94
N 26	458	452	386	385	310	322	242	279	207	271	408	318	228	165	132	445	354	251	149	85	79	405	297	191	94	**0**

The no mobility scenario with VoIP only profile (Figure 3-12), simulates a WMN where the nodes are communicating with each other, without any background traffic. Each node is talking to another node, simulating a VoIP conversation. Nodes are randomly selected according to Table 1. One node is acting as speaker 1, while the other node is acting as speaker 2. In each node, an STG (Send Traffic Grapher) service is running to send VoIP traffic and an RTG (Receive Traffic Grapher) service is running to receive VoIP traffic. Meanwhile, logs are collected during the VoIP conversations to analyze how the communication is proceeding, whether packet losses, jitters and delays are occurring and whether they are within the acceptable limits for voice communication. The VoIP only profile sends UDP traffic. VoIP conversations among the nodes start at varying intervals. Table 4 shows the time when the VoIP conversation starts.

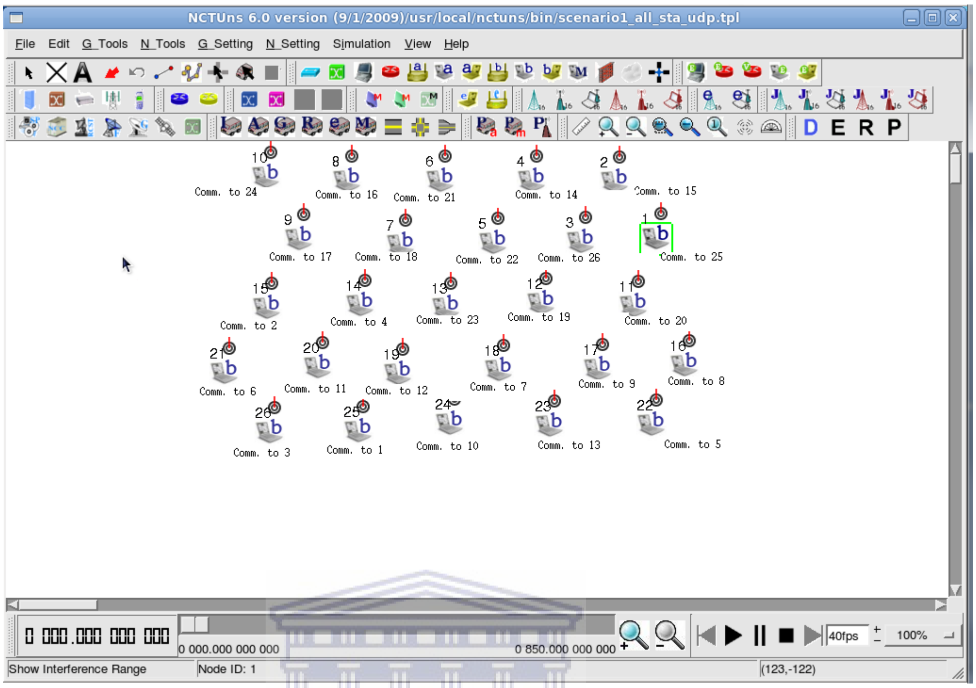


Figure 3-12 *No mobility, VoIP only profile*

According to Table 4, the RTG service starts once the simulation is run. Since the RTG service is only aimed at receiving the VoIP traffic generated by STG, this service must start before the STG starts sending traffic. This is why the start time for RTG is set to zero seconds. The STG services start one after the other and there is a difference of only one second between the starting time of each node's STG service.

Table 4 *Nodes VoIP Communication. (starting time in seconds)*

No	Speaker 1	Speaker 2	STG service Start time in sec	RTG service Start time in sec
1	Node 1	Node 25	1	0
2	Node 2	Node 15	2	0
3	Node 3	Node 26	3	0
4	Node 4	Node 14	4	0
5	Node 5	Node 22	5	0
6	Node 6	Node 21	6	0
7	Node 7	Node 18	7	0
8	Node 8	Node 16	8	0
9	Node 9	Node 17	9	0
10	Node 10	Node 24	10	0
11	Node 11	Node 20	11	0
12	Node 12	Node 19	12	0
13	Node 13	Node 23	13	0

A delay of 1 second has been configured, since in the real world, the VoIP communication does not start at the same time in all nodes. Each node might start its VoIP conversation at a different starting time. Therefore Node 1 starts sending VoIP traffic 1 second after the simulation starts, while Node 13 starts sending VoIP traffic 13 seconds after the simulation starts. As an example, the following commands are used in the simulation software to generate VoIP only traffic in Node 1, which is communicating with Node 25. All the remaining nodes are configured with the same commands.

Node 1 configuration:

stg -i spkrconfig1.cfg -p 4000 1.0.1.25

rtg -u -p 4000 -w pktlog1 -o thrlog1

Commands' Description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig1.cfg	*It is the configuration script file for speaker 1*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.25	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving (listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog1*
-o	*It writes the per packet throughput log into a file, named thrlog1*

Node 25 Configuration:

stg -i spkrconfig2.cfg -p 4000 1.0.1.1

rtg -u -p 4000 -w pktlog25 -o thrlog25

Commands description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig2.cfg	*It is the configuration script file for speaker 2*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.1	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*

-p	*It sets the receiving (listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog25*
-o	*Writes the per packet throughput log into a file, named thrlog25*

The no mobility scenario VoIP and non-VoIP profile is designed to simulate a VoIP conversation running, while background traffic is also transported by WMN. The VoIP traffic is generated according to Figure 3-9's configuration script. The non-VoIP traffic is generated using the STCP and RTCP packet generation tool, using the TCP greedy mode. Figure 3-13 shows a screenshot of the VoIP and non-VoIP profile simulation. .

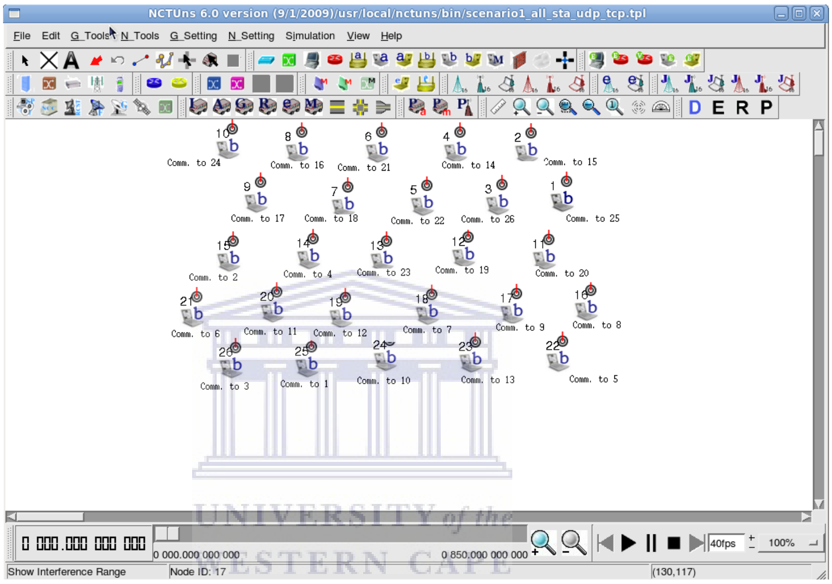


Figure 3-13 *VoIP and non-VoIP profile simulation*

As an example, when Node 1 communicates with Node 25, two types of traffic are transported between these two nodes. Node 1 is having a VoIP conversation with node 25 using the RTP/UDP/IP and meanwhile both nodes are having TCP connection for a file transfer from Node 25 to Node 1. The commands used to simulate this test case, are as follows:

Node 1 configuration:

stg -i spkrconfig1.cfg -p 4000 1.0.1.25

rtg -u -p 4000 -w pktlog1 -o thrlog1

rtcp -p 5000 -w rtcplog1

Commands Description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*

spkrconfig1.cfg	*It is the configuration script file for speaker 1*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.25	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving (listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog1*
-o	*It writes the per packet throughput log into a file, named thrlog1*
rtcp	*Receiving TCP traffic*
-p	*It sets the listening port to 5000*
-w	*It writes the per second throughput results in a file named rtcplog1*

Node 25 Configuration:

stg -i spkrconfig2.cfg -p 4000 1.0.1.1

rtg -u -p 4000 -w pktlog25 -o thrlog25

stcp -p 5000 1.0.1.1

Commands description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig2.cfg	*It is the configuration script file for speaker 2*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.1	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving (listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog25*
-o	*It writes the per packet throughput log into a file, named thrlog25*
stcp	*Sends TCP traffic in greedy mode*
-p	*Sets the destination port to 5000*
1.0.1.1	*The destination (receiving) node's IP address*

3.3.2 Limited mobility scenario

This scenario is designed and configured with 10 nodes, moving at a walking speed of 1.3m/sec, while the other 16 nodes are stationary. The aim of the scenario is to discover the effects of mobile nodes on VoIP traffic. The moving nodes' position information during the simulation time is shown in Table 5. The limited mobility scenario’s simulation design screen shot is shown in Figure 3-14.

Table 5 *Limited mobility scenario nodes’ movement information*

No	Node Name	Position/ Time	Coordinate Values (X , Y in metres and T in seconds)						
1	Node 2	X	271	310	692	642			
		Y	43	109	112	39			
		T	282.359	341.329	635.185	703.248			
2	Node 5	X	693	645	273	306	515		
		Y	106	37	43	107	112		
		T	141.591	206.247	492.438	547.828	708.644		
3	Node 8	X	274	315	697	650	359		
		Y	41	112	112	39	43		
		T	61.7113	124.779	418.625	485.411	709.278		
4	Node 9	X	691	647	277	308			
		Y	106	39	39	107			
		T	298.542	360.2	644.816	702.303			
5	Node 11	X	271	227	275	685	720	659	
		Y	181	250	310	304	235	175	
		T	297.86	360.81	419.916	735.335	794.849	860.667	
6	Node 13	X	270	230	280	690	720	659	457
		Y	181	254	313	310	241	169	176
		T	145.631	209.662	269.152	584.545	642.422	715.011	870.489
7	Node 16	X	662	271	230	280	691	720	
		Y	172	181	250	313	311	238	
		T	59.4899	360.339	422.079	483.948	800.106	860.528	
8	Node 21	X	280	692	721	666	272	231	
		Y	316	313	238	172	178	250	
		T	73.2387	390.17	452.025	518.112	821.224	884.959	
9	Node 23	X	682	721	662	272	233	284	583
		Y	302	235	170	181	250	317	311
		T	87.0897	146.724	214.25	514.369	575.338	640.108	870.155
10	Node 25	X	684	722	665	270	223	282	376
		Y	304	232	167	178	238	310	311
		T	244.626	307.251	373.753	677.717	736.345	807.95	880.262

The limited mobility scenario's VoIP only profile is designed in order to simulate a VoIP conversation between mesh nodes without any background traffic. As an example, Node 1 and Node 25 are carrying a VoIP conversation. The commands used to simulate this scenario are as follows:

Node 1 configuration:

stg -i spkrconfig1.cfg -p 4000 1.0.1.25

rtg -u -p 4000 -w pktlog1 -o thrlog1

Commands Description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig1.cfg	*It is the configuration script file for speaker 1*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.25	*It is the destination (receiver) IP address*

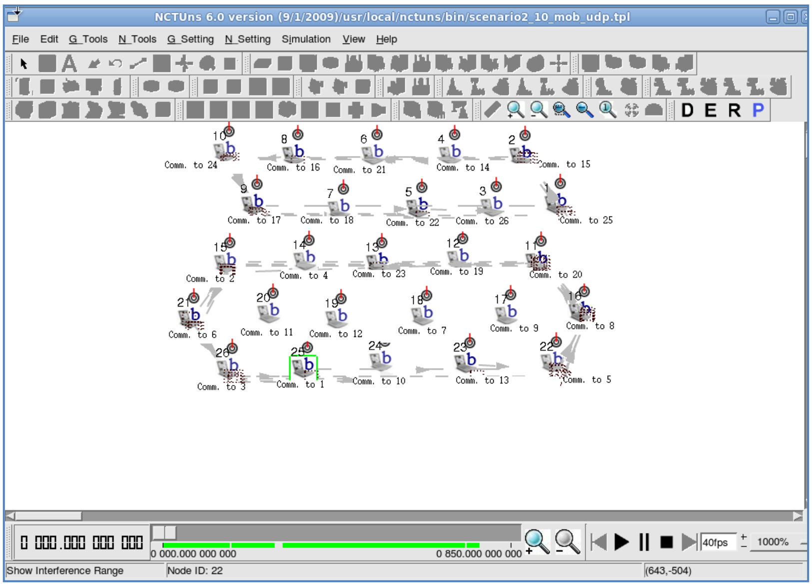


Figure 3-14 *Limited mobility scenario VoIP only profile*

rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog1*
-o	*It writes the per packet throughput log into a file, named thrlog1*

Node 25 Configuration:

stg -i spkrconfig2.cfg -p 4000 1.0.1.1

rtg -u -p 4000 -w pktlog25 -o thrlog25

Commands description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig2.cfg	*It is the configuration script file for speaker 2*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.1	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog25*
-o	*It writes the per packet throughput log into a file, named thrlog25*

The limited mobility scenario's VoIP with non-VoIP profile is designed to simulate a scenario where mesh nodes are carrying voice conversations and at the same time transporting non-VoIP traffic, while a number of mesh nodes are moving. In this scenario, 10 mesh nodes are configured to move at a walking distance of 1.3m/sec. As an example, when Node 1 communicates with Node 25, there are two types of traffic transported between these two mesh nodes. Node 1 is having a VoIP conversation with node 25 and transporting RTP/UDP/IP traffic and meanwhile both nodes are having a TCP connection for a file transfer from Node 25 to Node 1. The commands used to simulate this test case are as follows:

Node 1 configuration:

stg -i spkrconfig1.cfg -p 4000 1.0.1.25

rtg -u -p 4000 -w pktlog1 -o thrlog1

rtcp -p 5000 -w rtcplog1

Commands Description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig1.cfg	*It is the configuration script file for speaker 1*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.25	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog1*
-o	*It writes the per packet throughput log into a file, named thrlog1*
rtcp	*Receiving TCP traffic*
-p	*It sets the listening port to 5000*
-w	*It writes the per second throughput results in a file named rtcplog1*

Node 25 Configuration:

stg -i spkrconfig2.cfg -p 4000 1.0.1.1

rtg -u -p 4000 -w pktlog25 -o thrlog25

stcp -p 5000 1.0.1.1

Commands description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig2.cfg	*It is the configuration script file for speaker 2*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.1	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog25*
-o	*It writes the per packet throughput log into a file, named thrlog25*
stcp	*Sends TCP traffic in greedy mode*
-p	*It sets the destination port to 5000*
1.0.1.1	*The destination (receiving) node's IP address*

3.3.3 Full mobility scenario

This scenario is designed and configured to simulate all mesh nodes moving at a walking speed of 1.3m/sec. This scenario's node movement information is shown in Table 6. The mesh nodes are moving in horizontal and vertical paths, which are pre-defined. The network design screenshot is shown in Figure 3-15.

Table 6 ***Full mobility scenario's mesh node movement position information***

No	Name	Position /Time	Values (X, Y in metres, T in seconds)						
		X	642	273	309	690			
		Y	42	42	107	106			
1	**Node 1**	T	51.6932	336	393	685.774			
		X	273	309	687	642			
		Y	41	106	107	39			
2	**Node 2**	T	280.796	338	629	691.446			
		X	696	647	274	314	606		
		Y	106	38	39	112	109		
3	**Node 3**	T	72.4548	137	424	487.883	713		
		X	270	315	691	647	553		
		Y	41	106	109	35	42		
4	**Node 4**	T	215.434	276	565	631.712	704		
		X	695	640	270	317	514		
		Y	103	32	40	112	110		
5	**Node 5**	T	143.085	212	497	562.993	715		
		X	269	314	693	645	458		
		Y	39	110	109	37	41		
6	**Node 6**	T	141.572	206	498	564.337	708		
		X	698	653	274	316	417		
		Y	106	40	39	112	115		
7	**Node 7**	T	220.781	282	574	638.552	716		
		X	270	312	700	649	356		
		Y	42	107	110	35	41		
8	**Node 8**	T	64.8394	124	423	492.607	718		
		X	698	650	269	308			
		Y	106	37	40	106			
9	**Node 9**	T	303.925	369	662	720.638			
		X	310	700	647	271			
		Y	112	112	37	39			
10	**Node 10**	T	70.1857	370	441	730.064			
		X	269	233	283	685	719	661	
		Y	178	250	316	307	238	172	
11	**Node 11**	T	299.33	361	425	734.253	793	861	
		X	271	234	281	689	719	662	562
		Y	176	250	317	307	235	172	173
12	**Node 12**	T	222.469	286	349	663.006	723	788	865.286
		X	274	234	287	691	721	663	466
		Y	179	248	316	307	238	170	176
13	**Node 13**	T	142.476	204	270	580.992	639	708	859.228
		X	272	234	284	689	724	661	373
		Y	179	253	319	308	238	173	179
14	**Node 14**	T	74.4208	138	202	513.757	574	644	865.177
		X	232	283	699	727	665	280	
		Y	256	316	307	235	170	176	
15	**Node 15**	T	72.0166	133	453	512.091	581	877	
		X	664	272	228	279	688	712	
		Y	172	178	247	314	310	238	
16	**Node 16**	T	58.5071	360	423	487.802	802	861	
		X	322	632	322	623			
		Y	241	242	242	241			
17	**Node 17**	T	229.277	468	706	937.741			
		X	322	521	325	521	324	522	
		Y	244	242	244	241	247	244	
18	**Node 18**	T	148.589	302	452	603.238	755	907	
		X	645	410	645	417	648	414	
		Y	238	241	242	244	239	244	
19	**Node 19**	T	182.314	363	544	719.26	897	###	
		X	639	321	639	322			
		Y	238	239	239	241			
20	**Node 20**	T	243.107	488	732	976.19			
		X	287	699	721	670	273	238	
		Y	316	308	232	172	179	253	
21	**Node 21**	T	76.5298	394	454	514.948	820	883	
		X	725	666	279	239	284	684	
		Y	235	170	181	253	317	304	
22	**Node 22**	T	58.5071	126	424	487.203	547	855	
		X	694	727	674	276	233	282	574
		Y	307	232	173	178	253	316	304
23	**Node 23**	T	96.5959	160	221	526.811	593	655	879.512
		X	693	728	664	282	234	286	475
		Y	301	238	169	181	251	314	310
24	**Node 24**	T	177.03	232	305	598.853	664	727	872.397
		X	694	729	663	278	239	280	375
		Y	304	238	169	181	253	316	317
25	**Node 25**	T	252.318	310	383	679.531	743	800	873.42
		X	689	724	671	282	237	284	
		Y	305	235	172	178	256	316	
26	**Node 26**	T	320.778	381	444	743.575	813	871	

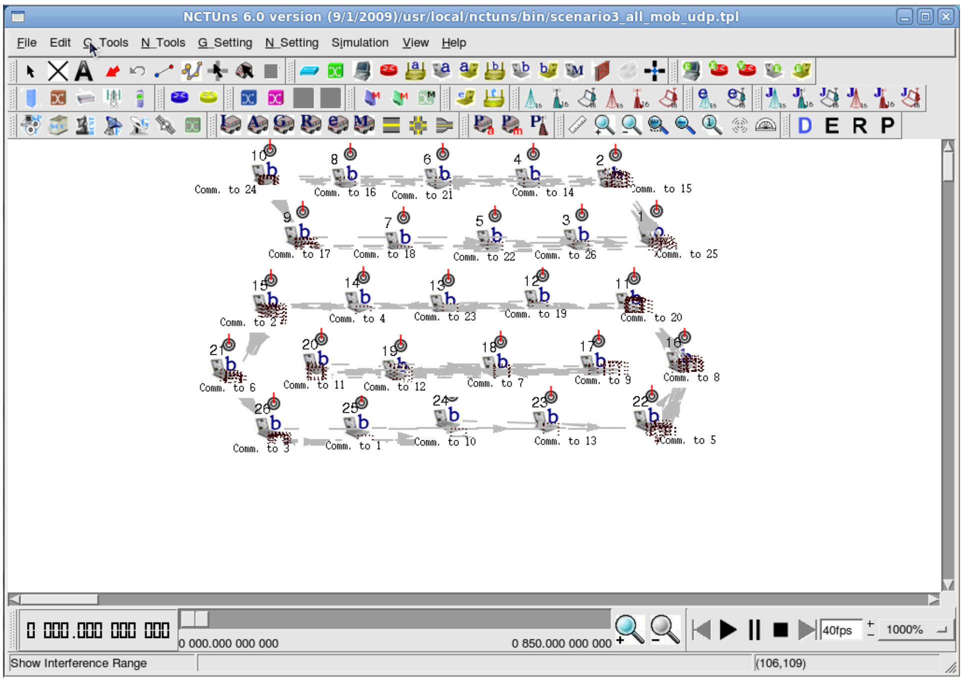


Figure 3-15 *Full mobility scenario*

The full mobility scenario's VoIP only profile is designed and configured in order to simulate a VoIP only profile, where mesh nodes are only involved in VoIP conversation, while the nodes are moving. As an example, Node 1 and Node 25 are carrying VoIP communication between each other. The commands used to simulate this scenario are as follows:

Node 1 configuration:

stg -i spkrconfig1.cfg -p 4000 1.0.1.25

rtg -u -p 4000 -w pktlog1 -o thrlog1

Commands Description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig1.cfg	*It is the configuration script file for speaker 1*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.25	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog1*
-o	*It writes the per packet throughput log into a file, named thrlog1*

Node 25 Configuration:

stg -i spkrconfig2.cfg -p 4000 1.0.1.1

rtg -u -p 4000 -w pktlog25 -o thrlog25

Commands description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig2.cfg	*It is the configuration script file for speaker 2*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.1	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog25*
-o	*It writes the per packet throughput log into a file, named thrlog25*

The full mobility scenario's VoIP and non-VoIP profile is designed and configured to simulate mesh nodes' VoIP conversations while the nodes are carrying non-VoIP traffic, by establishing a TCP session. All mesh nodes are moving at the same time. As an example, when Node 1 communicates with Node 25, there are two types of traffic exchanges between these two nodes. Node 1 and 25 are having a VoIP conversation and transporting RTP/UDP/IP while both nodes are having TCP sessions for a file transfer from Node 25 to Node 1. The commands used to simulate this scenario are as follows:

Node 1 configuration:

stg -i spkrconfig1.cfg -p 4000 1.0.1.25

rtg -u -p 4000 -w pktlog1 -o thrlog1

rtcp -p 5000 -w rtcplog1

Commands Description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig1.cfg	*It is the configuration script file for speaker 1*

-p	*It is an option to set the destination port, which is 4000*
1.0.1.25	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog1*
-o	*It writes the per packet throughput log into a file, named thrlog1*
rtcp	*Receiving TCP traffic*
-p	*It sets the listening port to 5000*
-w	*It writes the per second throughput results in a file named rtcplog1*

Node 25 Configuration:

stg -i spkrconfig2.cfg -p 4000 1.0.1.1

rtg -u -p 4000 -w pktlog25 -o thrlog25

stcp -p 5000 1.0.1.1

Commands description:

stg	*The send traffic grapher*
-i	*It sets the stg mode to configuration file*
spkrconfig2.cfg	*It is the configuration script file for speaker 2*
-p	*It is an option to set the destination port, which is 4000*
1.0.1.1	*It is the destination (receiver) IP address*
rtg	*Receive traffic grapher*
-u	*It sets the rtg mode to UDP mode*
-p	*It sets the receiving(listening) port to 4000*
-w	*It writes the packet log into a log file, named pktlog25*
-o	*It writes the per packet throughput log into a file, named thrlog25*
stcp	*It sends TCP traffic in greedy mode*
-p	*It sets the destination port to 5000*
1.0.1.1	*The destination (receiving) node's IP address*

3.4 Data collection

For data collection purposes, the traffic generator and simulation software capabilities were used to log the traffic generation results. The STG and RTG traffic generation tools have the capability to write the logs into a file. The RTG usage options are shown in Figure 3-16.

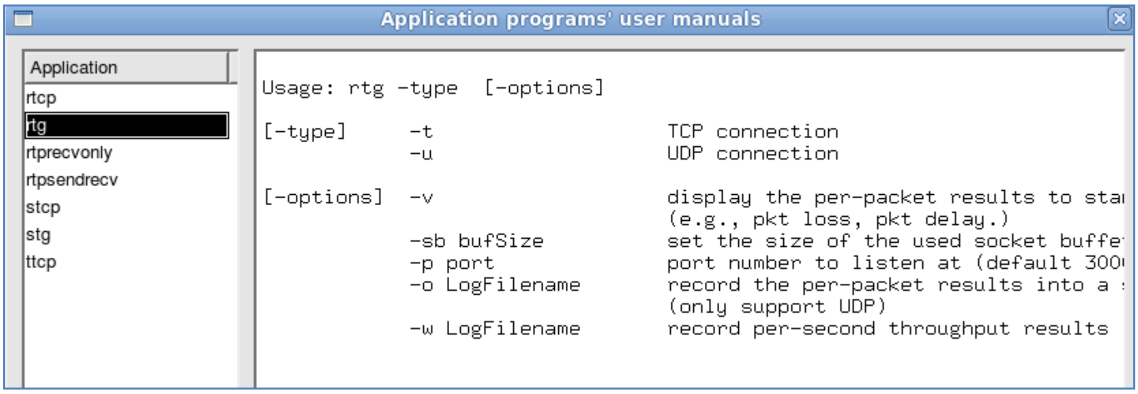


Figure 3-16 *RTG usage options*

In each mesh node the RTG service is running and the traffic that is sent by the STG, is logged in a text file. The RTG log file sample is shown in Figure 3-17. This log file shows the average packet loss and average delay.

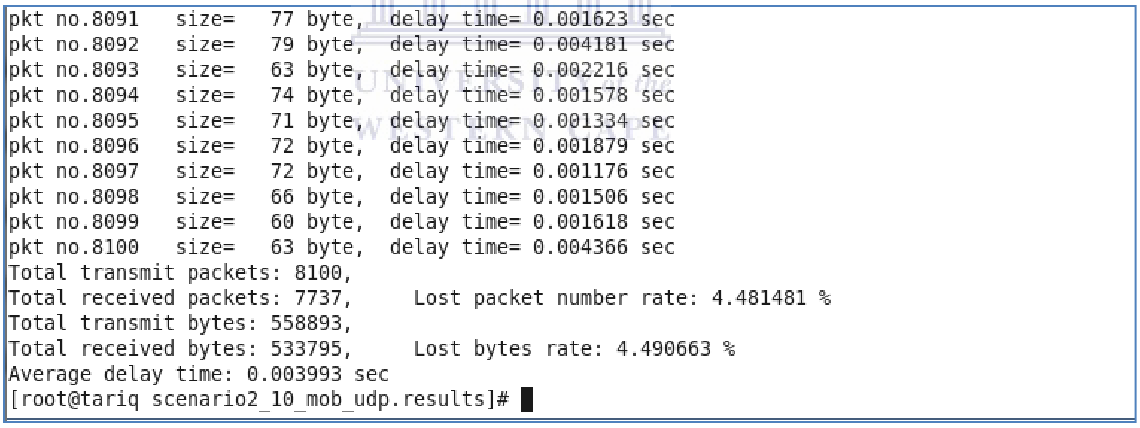


Figure 3-17 *RTG log file content*

The throughput logs are extracted from the NCTUns using its throughput logging capability. A log file is generated as shown in Figure 3-18. All the throughput log files are imported into an Excel file and then the average throughput is calculated in Kbps for each node, since NCTUns log the throughput in terms of Kbps.

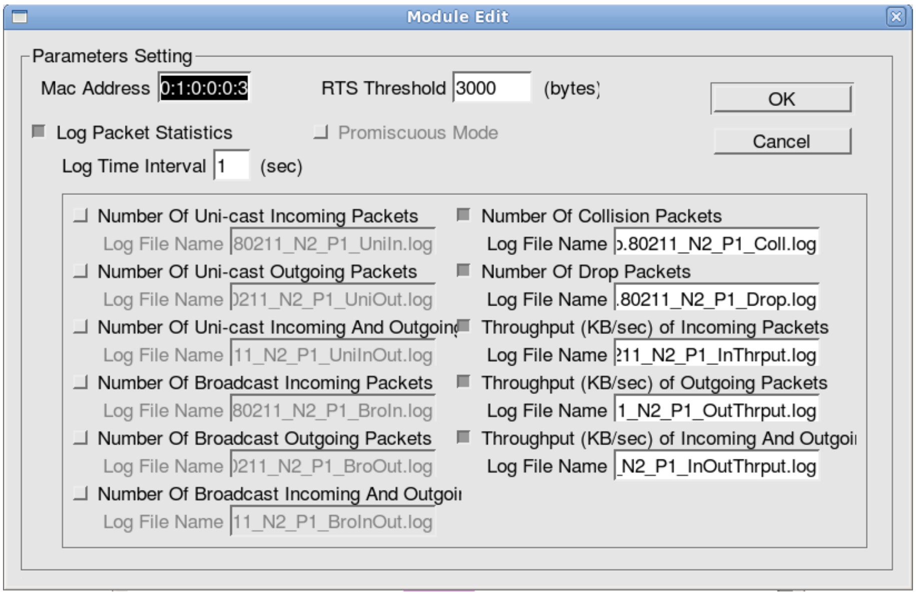


Figure 3-18 *Throughput logging in NCTUns*

Jitter rate is calculated from the packet log. Each packet is numbered in a sequence and is time-stamped. All the packet logs are separately imported for each node into an Excel file. Next, the packet log is sorted by sequence number. The time difference between the first and second packet received, is then calculated. This method is used for all the packets in the log file. In the end, the average time difference between all the packets is calculated.

3.5 Summary

This chapter presented the methods and steps used to run the three WMN scenarios. Each scenario was tested against two simulation cases or traffic profiles. The simulation cases reflected VoIP and non-VoIP traffic profiles. The three scenarios' simulation cases were designed and configured to simulate mesh nodes in stationary mode (No mobility), 10 mobile mesh nodes and 16 stationary nodes (Limited mobility) and all mesh nodes mobile (Full mobility). The VoIP profile was based on a simple voice conversation between a mother and a child, which was converted into a VoIP traffic profile, based on human conversation VoIP characteristics and patterns. The VoIP conversation duration was 562 seconds, which included the talk periods and the silent periods. During the talk period, VoIP packets were exchanged between the VoIP peers using STG and RTG traffic generation tools, and during the silent period no VoIP packets were sent. The non-VoIP traffic profile was simulated using STCP and RTCP in greedy mode. Data collection was done on the source and destination nodes using the STG and RTG and simulation software logging capabilities during the simulation process.

4 Analysis of results

The analysis of results is based on simulation scenarios that were used to collect the required logs, statistics and data from each of the mesh nodes. The NCTUns simulation software was configured to generate logs for analyzing the VoIP QoS factors of delay, jitter, packet loss and throughput. The STG and RTG tools packet and throughput logging options, along with mesh nodes' logging features, were used to collect such logs. Each of these factors is analyzed separately in this chapter. First, each scenario's delay, jitter, packet loss and throughput results were analyzed for each one of the traffic profiles and finally all scenarios were compared considering delay, jitter, packet loss and throughput results for all the traffic profiles. To present the analysis of the results, this chapter is structured as follows: Section 4.1 presents the analysis of no mobility scenario with each VoIP QoS factor analyzed separately. Section 4.2 contains an analysis of the VoIP QoS factors for the limited mobility scenario, with each factor analyzed separately. Section 4.3 presents the analysis results for the full mobility scenario. Again all the VoIP QoS factors are analyzed and discussed separately. Section 4.4 contains a synthesis of the results, with an analytical comparison of the three scenarios and with corresponding traffic profiles. Finally, Section 4.5 contains a summary of the analyses.

4.1 No mobility scenario results

The analysis of delay, jitter, packet loss and throughput in the no mobility scenario, shows whether the results of the simulation cases are in acceptable range for the VoIP QoS factors. According to the literature survey, the acceptable range for VoIP QoS factors is defined as less than 200ms delay, less than 100ms jitter, and up to 5% packet loss.

Delay analysis for VoIP only profile (No mobility UDP) shows that delay is very low and does not exceed the VoIP acceptable limit of less than 200ms, therefore the delay factor is not an issue in this scenario. The second profile, simulating the VoIP and non-VoIP traffic (No mobility UDP+TCP), shows that adding the TCP as the background traffic, affects the delay factor to a large extent. It also shows that delay increases and even exceeds the VoIP QoS limit of 200ms. In this test case, the delay even exceeds 2000ms. There are only 5 nodes that have a delay of less than 200ms.The reason is that these nodes are located close to each other, i.e. they are only one hop away. It is clear that mixing TCP traffic with VoIP applications, will not be a sensible implementation, as VoIP applications will become unusable.

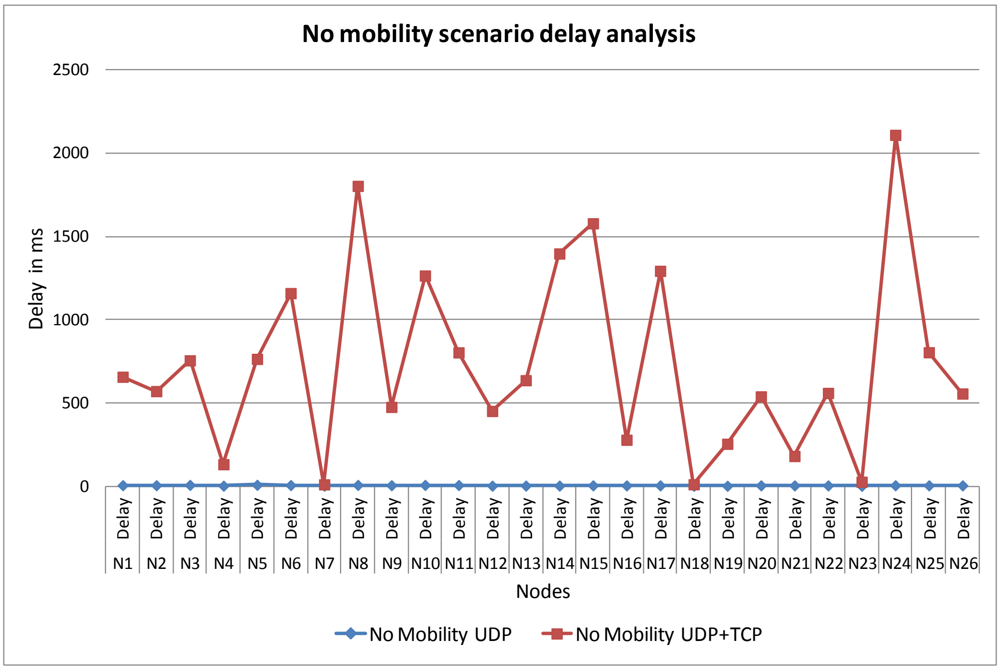


Figure 4-1 *No mobility scenario: Delay analysis*

Jitter analysis for the no mobility scenario is shown in Figure 4-2. The VoIP only profile was compared to the VoIP and non-VoIP profiles. The comparison shows that the VoIP only profile has a lower jitter rate than VoIP with non-VoIP profile, but both scenarios' results show that the jitter limits of VoIP are not exceeded. All the jitter rates are less than 100ms, which is the acceptable jitter rate for VoIP. Therefore jitter rate doesn't seem to be a big a concern in this scenario. The results show that VoIP implementation without the background traffic performs much better than having background traffic like TCP, but even so, the jitter rate does not cross the acceptable rate for VoIP.

The packet loss analysis for the no mobility scenario is shown in Figure 4-3. The two scenarios, VoIP-only profile, and VoIP and non-VoIP profiles, were compared. According to the graph, a VoIP only profile, where there is no background traffic, has almost no packet loss or very little loss compared to a VoIP profile mixed with background traffic. If background traffic is added, the packet loss rate increases and the graph shows that the packet loss rate reaches as much as almost 80%, while the acceptable packet loss rate for VoIP application is less than 5%. It can be concluded that packet loss rate with VoIP only profile is at acceptable range for VoIP traffic, but mixing VoIP traffic with non-VoIP traffic, renders the VoIP implementation unusable.

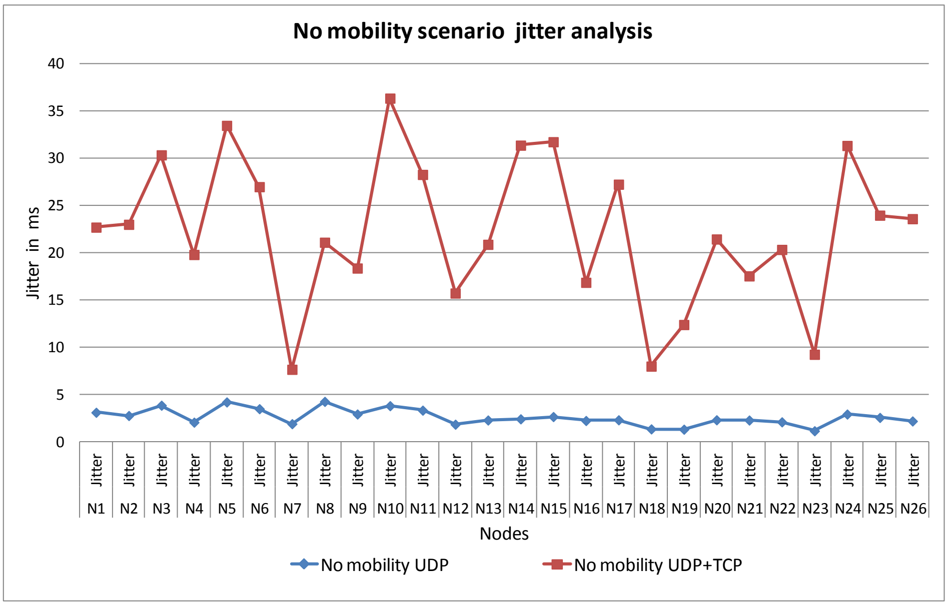


Figure 4-2 *No mobility scenario: Jitter analysis*

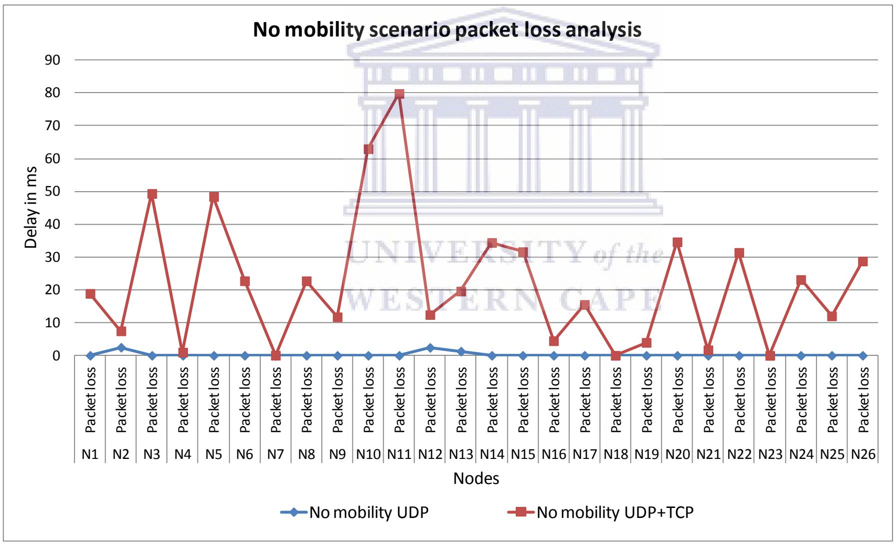


Figure 4-3 *No mobility scenario: Packet loss analysis*

The **throughput analysis** for the no mobility scenario is shown in Figure 4-4. Here a comparison of a VoIP only profile with VoIP and non-VoIP profile is done. As the graph shows, the VoIP only profile requires very low bandwidth to transport VoIP packets, since VoIP packets are very small. A simple calculation shows that for a normal VoIP conversation with G.729 codec, 50 packets per second have to be sent. Each packet will have an average size of 70 bytes, which will contain the RTP/UDP/IP data and headers. If the maximum packet size of 80 bytes is considered, then 4000 bytes per second have to be sent. To send 4000 bytes per second, a bandwidth of 32 kbps is required. If it is roughly calculated, and layer two encapsulation is considered, together with a bandwidth of 35-

40kbps, it will be enough for a successful VoIP conversation. As the graph shows, the VoIP only profile (No mobility UDP) performs well and it uses as much bandwidth as required by the VoIP application. The bandwidth usage is around 40-45 kbps. Looking at the VoIP and non-VoIP profile, with the TCP as the background traffic, the bandwidth usage increases. Once TCP starts sending and exchanging traffic, bandwidth is allocated to TCP traffic and there are chances that lower bandwidth is allocated to VoIP application/traffic. Although the bandwidth usage is high, VoIP traffic still uses normal packet queues and is mixed with TCP traffic. This increases delay and jitter and even packet loss. Since the packets are delayed, they will be timed out and will be of no use by the VoIP applications. It can be seen in the graph that the VoIP only profile, uses lower bandwidth, but performs well, since the bandwidth is not shared with other non-VoIP applications.

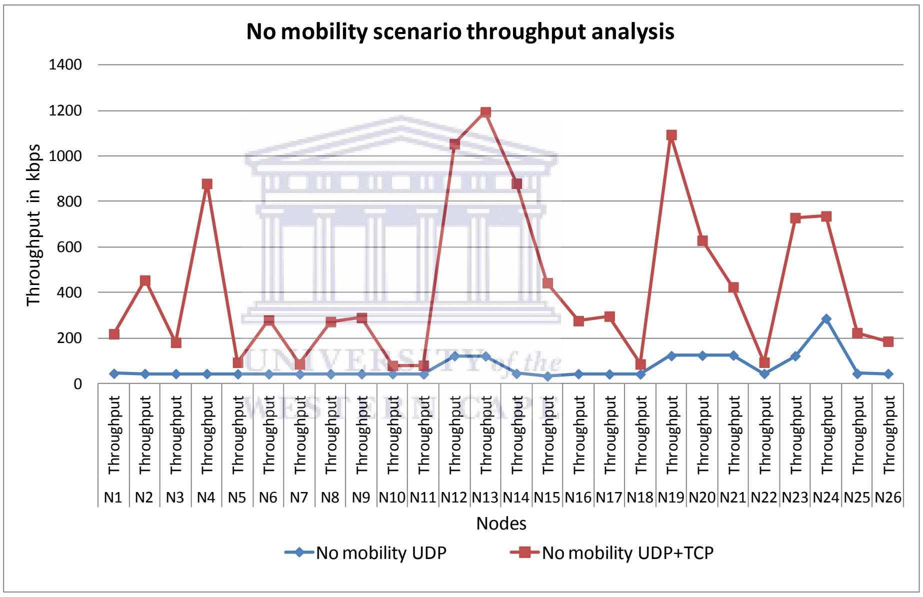


Figure 4-4 *No mobility scenario: Throughput analysis*

4.2 Limited mobility scenario analysis

As discussed in Section 3.3.2, only 10 mesh nodes were configured to move at a walking speed of 1.3m/sec. In this scenario, nodes 2, 5, 8, 9, 11, 13, 16, 21, 23 and 25 are moving. Each one of the VoIP QoS factors like delay, jitter, packet loss and throughput are analyzed. This scenario is set up in two separate profiles. One is a VoIP only profile and another one is a VoIP and non-VoIP traffic profile. In this scenario both traffic profiles are analyzed and their effect on VoIP QoS factors is investigated.

Delay analysis for VoIP only profile is shown in Figure 4-5. This analysis show that VoIP only profile (Limited mobility UDP) performs well and the delay does not exceed the acceptable limit of less than 200ms. Although some of the mesh nodes' delay increased due to the mobility factor, the delay still falls in the acceptable range for VoIP. It is clear that the delay factor is affected by node mobility. It can also be seen that the VoIP only profile performs well and VoIP applications can run successfully in these scenarios.

If the VoIP traffic is mixed with non-VoIP traffic (Limited mobility UDP and TCP), it shows that there is a considerable variation in the delay time. Delay exceeds the acceptable VoIP limit of less than 200ms, and even reaches 2000ms in one case. As in a previous scenario, nodes 7, 8 and 19 again experience delay of less than 200ms. The reason is that these nodes are only one hop away from their communication nodes. These 3 nodes' delay results show that although VoIP and non-VoIP traffic is mixed, the delay is still in an acceptable range for VoIP, since these nodes are not multiple hops away. It is also observed that nodes 6, 8, 11, 16, 15, 17, 20 have lower delay than in the no mobility UDP and TCP scenario, because their communication nodes are coming closer as the nodes move.

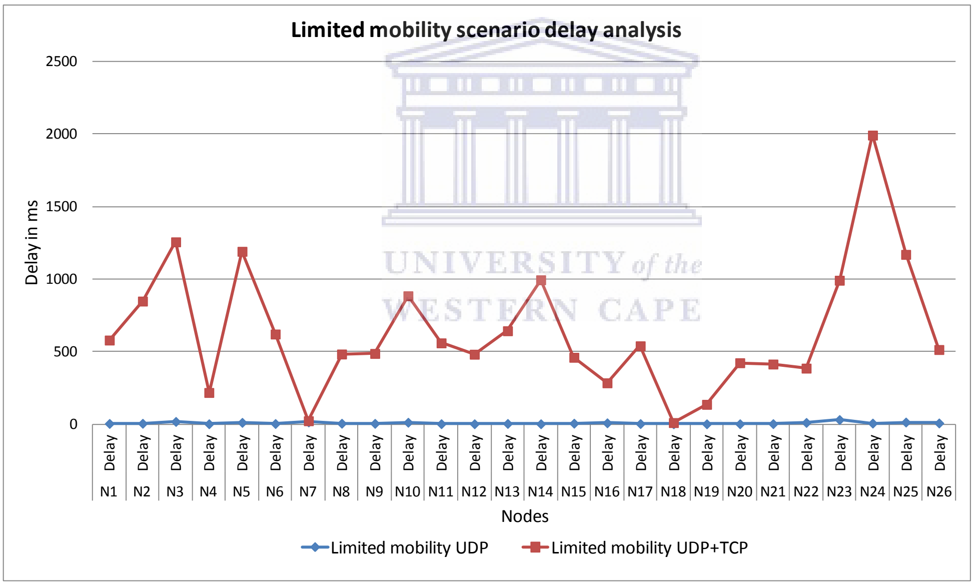


Figure 4-5 *Limited mobility scenario: Delay analysis*

Jitter analysis for the limited mobility scenario is shown in Figure 4-6. According to the achieved results from limited mobility simulation and VoIP only profile with VoIP and non-VoIP profile, it is evident that there is much difference in the jitter values of these two profiles. Even so, the jitter values on both profiles do not cross the acceptable jitter limit for VoIP of less than 100 msec. In summary, the jitter factor will not be of much concern in VoIP implementation in both traffic profiles. Also, these values are not much different than the no mobility scenario for VoIP and non-VoIP profiles.

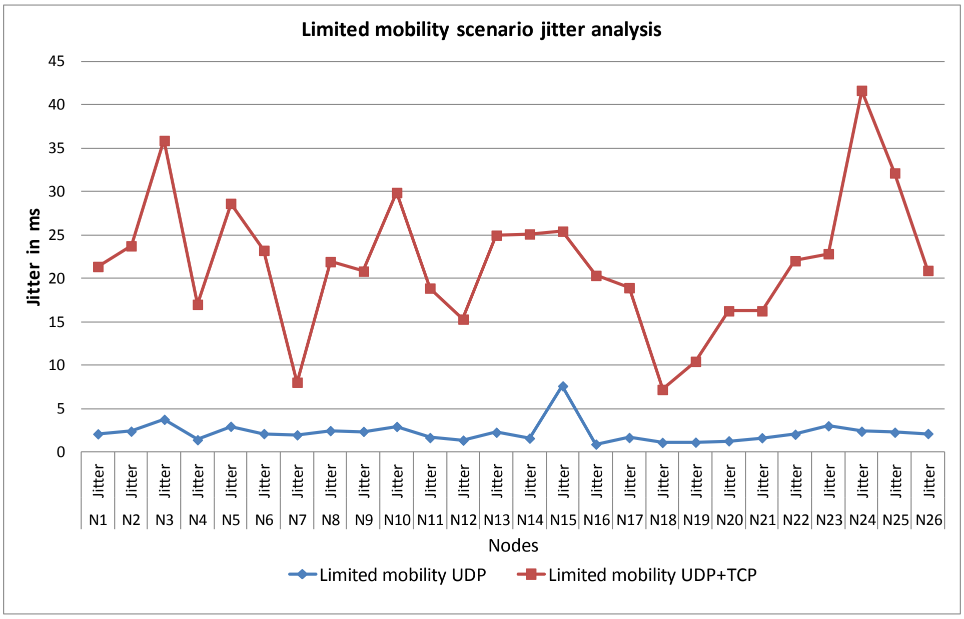


Figure 4-6 *Limited mobility scenario: Jitter analysis*

Packet loss analysis for the limited mobility scenario is shown in Figure 4-7. It is evident that in the VoIP only profile, nodes that are moving are affected by greater packet loss. The packet loss can either be seen on the moving nodes themselves or their associated hosts, with which they are communicating. Table 7 shows the communicating peers, and from these peers only nodes 2, 5, 8, 9, 11, 13, 16, 21, 23 and 25 are moving. The packet loss graph shows that nodes 2, 5, 8, 9 and 11 which are moving, experience packet loss, and nodes 1 and 6 also experience packet loss, since their associated communicating nodes, 25 and 21 respectively, are also moving. It is also observed that nodes 3, 7 and 10 experience packet loss, due to the fact that the nodes through which they are communicating, nodes 13, 21 and 25, are moving. The graph also shows that nodes 2, 5 and 8 crossed the acceptable packet loss rate of 5% for VoIP applications.

Table 7 *Limited mobility nodes' communication peers*

Speaker 1	Node 1	Node 2	Node 3	Node 4	Node 5	Node 6	Node 7	Node 8	Node 9	Node 10	Node 11	Node 12	Node 13
Speaker 2	Node 25	Node 15	Node 26	Node 14	Node 22	Node 21	Node 18	Node 16	Node 17	Node 24	Node 20	Node 19	Node 23

In the VoIP and non-VoIP traffic profile simulation results, a huge packet loss can be seen, compared to VoIP only profile. This is due the TCP background traffic. Most of the nodes have crossed the acceptable packet loss rate of 5%. Only nodes 4, 7, 12 and 18 have a less than 5% packet loss and this is due to the fact that these nodes are communicating with the hosts that are only one hop away, or are located closer. From this analysis, it can be concluded that mobility results in packet loss. In a VoIP only profile, the packet loss is mostly at an acceptable rate, although in some cases, it can be unacceptable. Also, mixing non-VoIP traffic with VoIP traffic can greatly affect VoIP conversation, which makes the VoIP implementation in these environments unusable, despite the fact that a few nodes which are one hop away, or located close to their VoIP peers, can have packet loss at acceptable rates.

The **throughput analysis** of the limited mobility scenario profiles is shown in Figure 4-8. The graph shows that the VoIP only profile uses very low throughput, since the VoIP packets are very small. Some nodes, where their communicating nodes are moving and in the process moving closer to their peers, can benefit from higher bandwidth. The graph also shows that nodes 20, 21, 23, 24 and 25 are using higher bandwidth, since during the movement process, they come closer to their communicating nodes. Node 24 is also benefitting from higher bandwidth, since its routing node, node 24, is moving closer at one point and then moving closer to node 10, which is communicating with node 24, at another point of the movement.

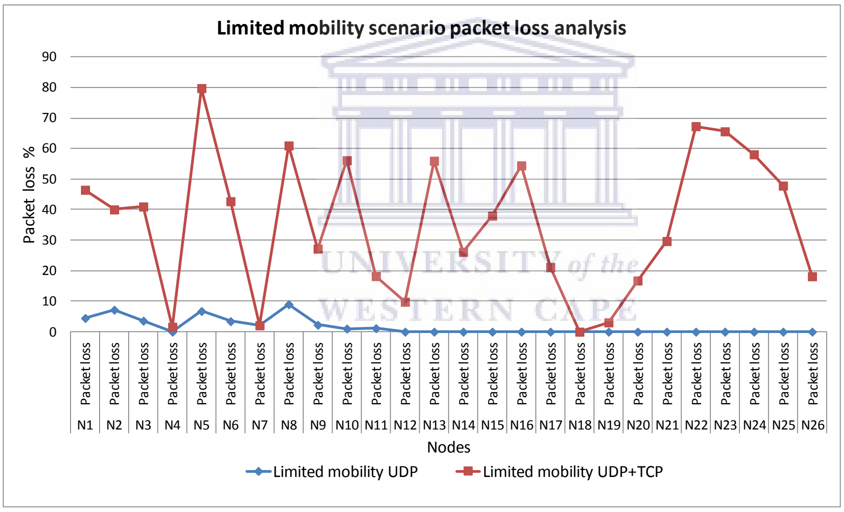


Figure 4-7 *Limited mobility scenario: Packet loss analysis*

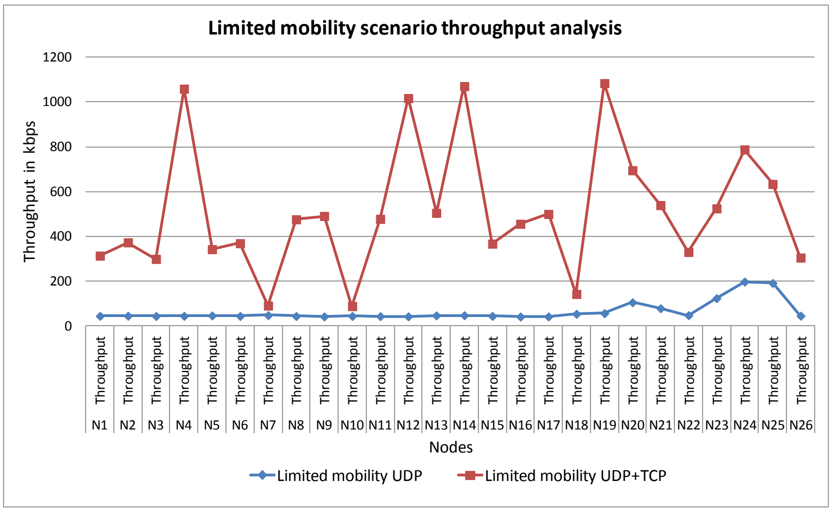


Figure 4-8 *Limited mobility scenario: Throughput analysis*

The profile for VoIP traffic mixed with non-VoIP, shows that throughput usage reaches a very high level, since TCP is using most of the available bandwidth. The way resource allocation is done for TCP and UDP, can result in less throughput allocation to VoIP traffic, which can result in further delay, jitter and packet loss. Only nodes which are located close to each other, or a hop away, can benefit from higher throughputs of above 800 Kbps, while the rest of the nodes are allocated lower throughputs. It can be concluded that in VoIP only profile, all of the throughput is allocated to VoIP traffic. Therefore it performs well and uses as much bandwidth as required, but usually bandwidths of 40-45 Kbps are enough for a successful VoIP conversation. Looking at the VoIP and non-VoIP profile mixed traffic patterns, it is evident that most of the bandwidth is used by the non-VoIP profile. Since there is no QoS mechanism implemented among the mesh nodes, VoIP traffic uses the normal queues, which can result in delay, jitter and packet loss. The analysis of delay, jitter and packet loss, also confirms this problem.

4.3 Full mobility scenario analysis

In the full mobility scenario, all mesh nodes are moving at a walking speed of 1.3m/sec. This scenario consists of two profiles. One is a VoIP only profile, where all mesh nodes are only involved in VoIP conversations and there is no background traffic. The second is a profile mixed with VoIP and non-VoIP traffic. An analysis of VoIP QoS factors, delay, jitter, packet loss and throughput, proves that they are exceeding acceptable ranges.

A **delay analysis** for the full mobility scenario is shown in Figure 4-9. The graph shows that delay for VoIP only profile is not crossing the VoIP delay limit of 200ms. Movement information of nodes is shown in Table 6 on page42. Referring to the second profile in which VoIP traffic is mixed with non-VoIP traffic, it can be seen that the delay exceeds the acceptable limit of 200ms. The delay reaches as high as about 1200 seconds. Only node 7 is an exception. It has a delay of around 21ms, which is due to its close proximity to its communicating host, which is node 18, and which is only one hop away. It can be concluded that if nodes are mobile and they are only carrying VoIP traffic and no background traffic exists, then VoIP implementations can be successful. But if nodes are carrying a mix of VoIP and non-VoIP traffic, it will make the VoIP applications unusable.

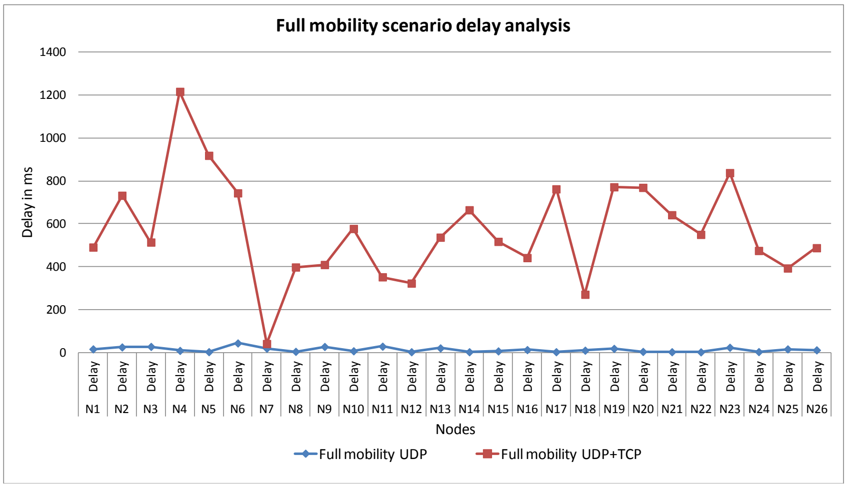


Figure 4-9 *Full mobility scenario: Delay analysis*

Jitter analysis for the full mobility scenario is shown in Figure 4-10. As the graph shows, jitter values of the VoIP only profile falls within the acceptable jitter limit of VoIP, which is less than 100ms. Looking at the jitter rates of VoIP, mixed with non-VoIP traffic, it can be seen that the jitter rates are higher. Despite the higher jitter rates in the second traffic profile, it still does not cross the VoIP jitter limit. In conclusion, the node mobility factor doesn't affect the jitter rate to the extent that would make the VoIP applications unusable, and although the non-VoIP traffic injection increases the jitter rates, it doesn't exceed the acceptable limits.

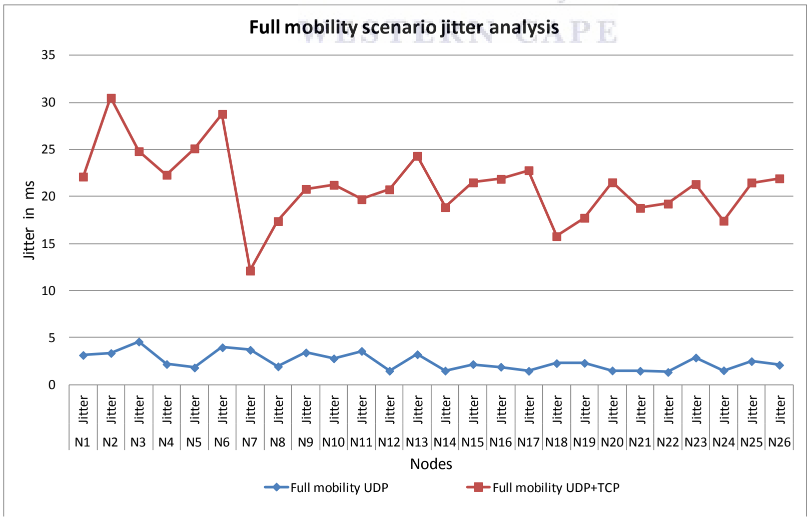


Figure 4-10 *Full mobility scenario: Jitter analysis*

Packet loss analysis for the full mobility scenario is shown in Figure 4-11. The graph shows that in the VoIP only profile, packet loss rate in the nodes where the communicating node is moving away from its peer, is higher than in those nodes where their communicating nodes are not moving very far from each other. These types of nodes have also crossed the acceptable packet loss rate for VoIP, which is less than 5%. The graph also shows that nodes 3, 4, 7, 10, 11, 12, 13 and 23 experience a packet loss rate of more than 5%, while the other nodes' packet loss rate is acceptable. Looking at the VoIP traffic mixed with non-VoIP traffic profile, it can be seen that packet loss rates increase dramatically due to the non-VoIP traffic existence, which is running as the background traffic. The result is that all nodes experience packet loss rates of above 5%. According to above analysis, it can be concluded that VoIP applications in scenarios where all nodes are mobile and mesh nodes are only exchanging VoIP packets, can be 70% successful, considering the packet loss factor. Unfortunately, VoIP applications in scenarios where VoIP traffic is mixed with non-VoIP traffic are completely unusable as a result of the packet loss factor.

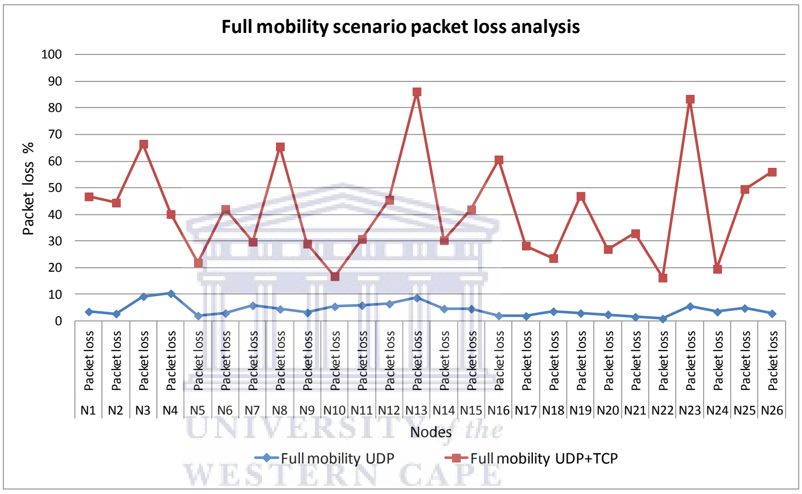


Figure 4-11 *Full mobility scenario: Packet loss analysis*

Throughput analysis for the full mobility scenario is shown in Figure 4-12. The graph shows that mesh nodes in the VoIP only profile, only uses very small portions of the bandwidth required to make a VoIP call, which requires between 40-45 Kbps of bandwidth, because VoIP packets are very small. Referring to VoIP traffic mixed with non-VoIP traffic, it can be seen that bandwidth usage rises in most of the nodes and nodes start dedicating bandwidth for the TCP connections. This results in allocating lower bandwidth than the VoIP requirement. And besides, the VoIP packets will be using the same queues as other non-VoIP traffic, which can result in more delay, jitter and packet loss. This happens because there is no QoS mechanism used in the mesh nodes to tag VoIP traffic and to place them in the priority queues. In conclusion, VoIP applications perform well considering bandwidth usage, providing that they are not mixed with other types of traffic, but if they are mixed with other types of traffic, QoS mechanisms should be implemented among the nodes.

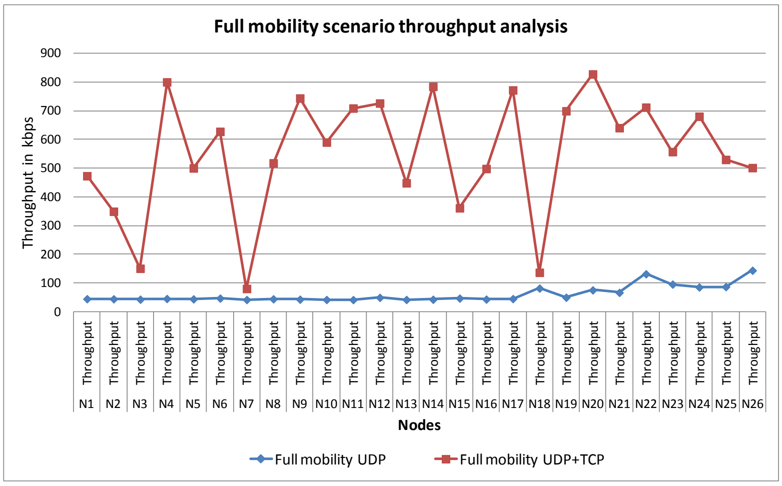


Figure 4-12 *Full mobility scenario: Throughput analysis*

4.4 Comparative analysis

In this section the different scenarios' results are compared to analyse the effect of nodes' mobility on VoIP QoS. The comparison basis will be QoS factors of delay, jitter, packet loss and throughput. Each QoS factor and each node's behaviour are compared against all scenarios. The focus will be on analyzing the effects of limited mobility and full mobility on the QoS factors, considering the VoIP and non-VoIP traffic profiles.

Delay analysis and a comparison of the three scenarios of VoIP, with its related traffic profiles, are shown in Figure 4-13. As the graph shows, VoIP only profiles without background traffic, experience delay lower than 200ms, while scenarios with background traffic, experience higher delays. The graph also shows that the VoIP only profile's delay values in the three scenarios are not much different, irrespective of the nodes' mobility factor.

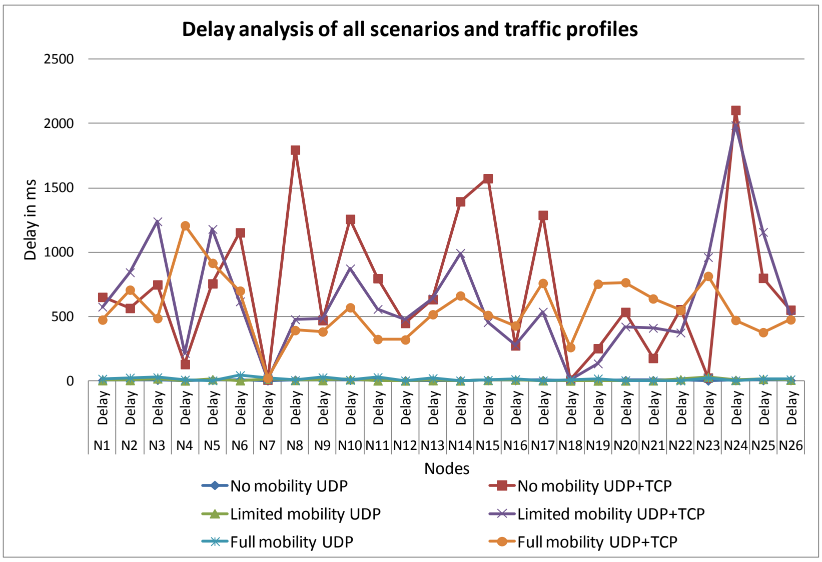


Figure 4-13 *Delay analysis of all scenarios and traffic profiles*

Jitter analysis of all three scenarios and their related traffic profiles are shown in Figure 4-14. The graph shows that the jitter rates of the VoIP-only profile, with no background traffic, are lower in all the scenarios, but the jitter rate for VoIP and non-VoIP profile is higher. It can also be seen that in all these scenarios, irrespective of the traffic profiles and nodes' mobility, the jitter rate is below the acceptable limit of 100ms.

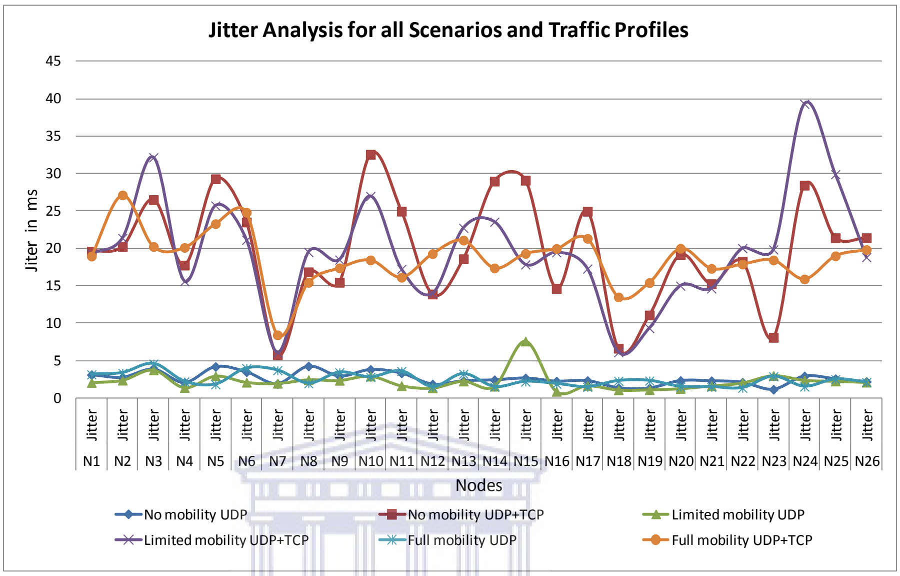


Figure 4-14 *Jitter analysis of all scenarios and traffic profiles*

Packet loss analysis of all the scenarios and their related traffic profiles, are shown in Figure 4-15. The graph shows that VoIP only profiles, with no background traffic, have lower packet loss rates than VoIP profiles with background traffic. Among all the scenarios, the no mobility scenario has the lowest packet loss rate, followed by the limited mobility scenario, with 10 mobile nodes, and the full mobility scenario, with all mobile nodes. The graph shows that node mobility results in packet loss. In the scenarios with background traffic, the full mobility scenario with the highest mobility rate has the highest packet loss rate as well. There are only a few mesh nodes in the limited mobility and full mobility scenarios that have lower packet loss. This is due to the fact that these nodes are communicating with the nodes that are only one hop away. Even when these nodes move, these values remain the same, since node movement is designed in a way that the nodes move one after the other, in a linear manner.

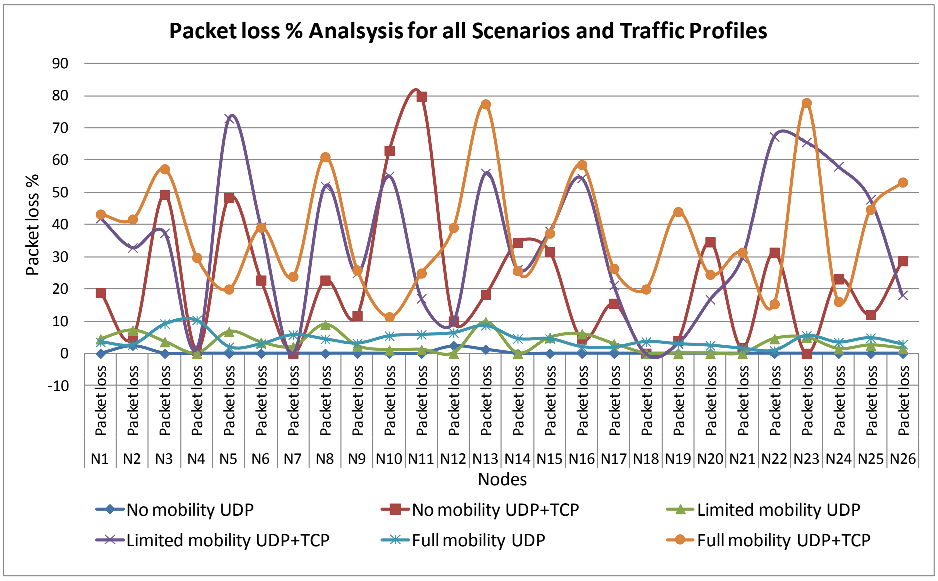


Figure 4-15 *Packet loss % of all scenarios and traffic profiles*

Throughput analysis of all three scenarios with their related traffic profiles, are shown in Figure 4-16. The comparative analysis shows that throughput usage in no mobility, limited mobility and full mobility with VoIP only profile, is between 40 and 45kbps. The graph shows that scenarios with background traffic have higher throughputs. Since no QoS mechanism is implemented on the mesh nodes, the VoIP and non-VoIP traffic share the same queues, which results in delay, jitter and packet loss.

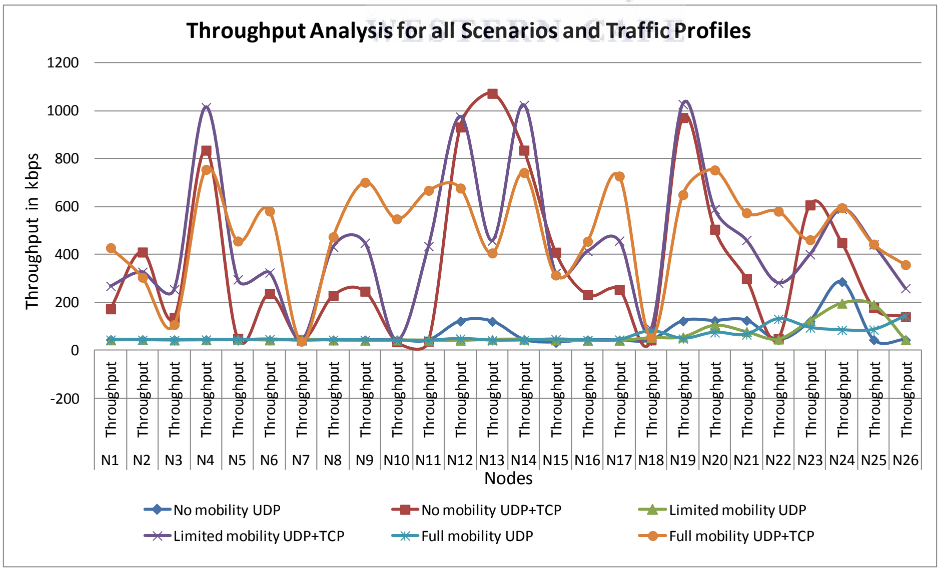


Figure 4-16 *Throughput analysis of all scenarios and traffic profiles*

A summary of all VoIP QoS factors, with the achieved results and their associated values for all simulation scenarios and profiles, is shown in Table 8.

Table 8 *Summary of all VoIP factors for all scenarios and traffic profiles*

Node Name	Factor	No mobility		Limited mobility		Full mobility	
		UDP	UDP+TCP	UDP	UDP+TCP	UDP	UDP+TCP
Node 1	Delay (ms)	5	651	4	576	16	475
	Packet loss %	0	19	4	42	4	43
	Throughput Kbps	45	174	44	269	44	429
	Jitter (ms)	3	20	2	19	3	19
Node 2	Delay (ms)	5	565	5	843	24	708
	Packet loss %	2	5	7	33	3	42
	Throughput Kbps	44	410	45	328	44	305
	Jitter (ms)	3	20	2	21	3	27
Node 3	Delay (ms)	7	748	18	1239	27	487
	Packet loss %	0	49	4	37	9	57
	Throughput Kbps	44	137	44	255	43	108
	Jitter (ms)	4	27	4	32	5	20
Node 4	Delay (ms)	2	129	2	216	8	1208
	Packet loss %	0	1	0	2	10	30
	Throughput Kbps	44	835	44	1015	45	755
	Jitter (ms)	2	18	1	16	2	20
Node 5	Delay (ms)	8	756	11	1180	3	916
	Packet loss %	0	48	7	73	2	20
	Throughput Kbps	43	51	46	296	44	456
	Jitter (ms)	4	29	3	26	2	23
Node 6	Delay (ms)	6	1153	3	618	43	700
	Packet loss %	0	23	3	39	3	39
	Throughput Kbps	43	237	44	325	47	582
	Jitter (ms)	3	24	2	21	4	25
Node 7	Delay (ms)	3	8	14	8	19	22
	Packet loss %	0	0	2	0	6	24
	Throughput Kbps	42	44	47	43	42	38
	Jitter (ms)	2	6	2	6	4	8
Node 8	Delay (ms)	7	1796	5	478	4	393
	Packet loss %	0	23	9	52	4	61
	Throughput Kbps	43	229	44	432	44	474
	Jitter (ms)	4	17	2	19	2	15
Node 9	Delay (ms)	5	470	5	483	27	382
	Packet loss %	0	12	2	25	3	26
	Throughput Kbps	43	247	42	449	43	702
	Jitter (ms)	3	15	2	18	3	17
Node 10	Delay (ms)	7	1258	12	873	7	570
	Packet loss %	0	63	1	55	5	11
	Throughput Kbps	44	36	44	44	42	549
	Jitter (ms)	4	33	3	27	3	18
Node 11	Delay (ms)	6	797	2	558	28	324
	Packet loss %	0	80	1	17	6	25
	Throughput Kbps	42	39	43	435	41	668
	Jitter (ms)	3	25	2	17	4	16
Node 12	Delay (ms)	2	449	2	480	2	320
	Packet loss %	2	10	0	10	6	39
	Throughput Kbps	122	932	42	975	49	678
	Jitter (ms)	2	14	1	14	1	19
Node 13	Delay (ms)	3	633	4	640	20	516
	Packet loss %	1	18	10	56	9	77
	Throughput Kbps	121	1073	46	458	42	407
	Jitter (ms)	2	19	2	23	3	21
Node 14	Delay (ms)	3	1394	2	993	2	662
	Packet loss %	0	34	0	26	5	26
	Throughput Kbps	45	836	47	1024	42	742
	Jitter (ms)	2	29	2	24	1	17
Node 15	Delay (ms)	4	1574	6	455	6	511
	Packet loss %	0	32	5	38	5	37
	Throughput Kbps	33	410	44	323	47	315
	Jitter (ms)	3	29	8	18	2	19
Node 16	Delay (ms)	4	274	8	276	12	429
	Packet loss %	0	4	6	54	2	59
	Throughput Kbps	44	233	41	415	43	455
	Jitter (ms)	2	15	1	19	2	20
Node 17	Delay (ms)	4	1289	3	537	2	760
	Packet loss %	0	16	3	21	2	26
	Throughput Kbps	42	254	42	458	45	728
	Jitter (ms)	2	25	2	17	1	21
Node 18	Delay (ms)	2	9	1	9	9	261
	Packet loss %	0	0	0	0	4	20
	Throughput Kbps	42	44	53	89	81	55
	Jitter (ms)	1	7	1	6	2	13
Node 19	Delay (ms)	2	252	1	134	18	754
	Packet loss %	0	4	0	3	3	44
	Throughput Kbps	123	972	56	1028	50	650
	Jitter (ms)	1	11	1	9	2	15
Node 20	Delay (ms)	4	535	2	421	4	765
	Packet loss %	0	35	0	17	2	25
	Throughput Kbps	124	506	105	590	75	753
	Jitter (ms)	2	19	1	15	2	20
Node 21	Delay (ms)	4	177	3	411	2	639
	Packet loss %	0	2	0	30	2	31
	Throughput Kbps	125	300	78	461	66	574
	Jitter (ms)	2	15	2	15	1	17
Node 22	Delay (ms)	4	556	11	375	2	548
	Packet loss %	0	31	5	67	1	15
	Throughput Kbps	44	50	47	283	131	581
	Jitter (ms)	2	18	2	20	1	18
Node 23	Delay (ms)	2	24	31	961	23	815
	Packet loss %	0	0	5	66	6	78
	Throughput Kbps	122	607	123	401	95	462
	Jitter (ms)	1	8	3	20	3	18
Node 24	Delay (ms)	5	2105	6	1987	3	472
	Packet loss %	0	23	1	58	3	16
	Throughput Kbps	287	450	197	591	85	595
	Jitter (ms)	3	28	2	39	2	16
Node 25	Delay (ms)	4	800	14	1157	15	377
	Packet loss %	0	12	3	48	5	45
	Throughput Kbps	45	179	191	442	87	443
	Jitter (ms)	3	21	2	30	2	19
Node 26	Delay (ms)	4	552	7	507	11	476
	Packet loss %	0	29	2	18	3	53
	Throughput Kbps	44	142	45	260	144	358
	Jitter (ms)	2	21	2	19	2	20

4.5 Summary

In this chapter all the simulation case results were analyzed individually and the traffic profiles within the same and different WMN scenarios, were also compared. . The use of self explanatory graphs show that VoIP applications' successful implementation can be achieved by isolating VoIP traffic to a separate network, where the resources are not shared with non-VoIP applications. The findings show that using non-VoIP applications along with VoIP applications affects all the VoIP QoS factors and makes the VoIP applications unusable. Table 9 shows the number of nodes exceeding the acceptable VoIP QoS limits in each scenario and each traffic profile. It also shows that jitter has remained below 100 ms in all scenarios, while VoIP only profile's delay and packet loss, have crossed the acceptable limits in a few nodes, which is due to nodes' mobility in the limited and full mobility scenarios. It is noticeable in both the VoIP and non-VoIP profiles that most of the nodes experience delay and packet loss higher than the acceptable ranges, which is due to the existence of the non-VoIP profile.

Table 9 *Number of nodes exceeding the acceptable VoIP QoS limits*

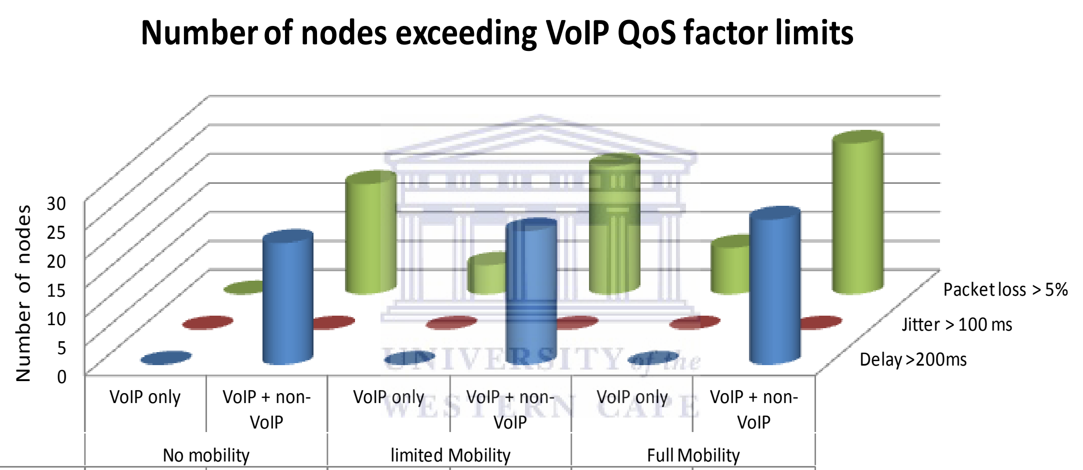
Scenario	Traffic Profile	Delay >200 ms	Jitter > 100 ms	Packet loss > 5%	Total No. of nodes
No mobility	VoIP only	0	0	0	26
	VoIP + non-VoIP	21	0	19	26
limited Mobility	VoIP only	0	0	5	26
	VoIP + non-VoIP	23	0	22	26
Full Mobility	VoIP only	0	0	8	26
	VoIP + non-VoIP	25	0	26	26

5 Conclusion and future work

In this research, the focus was to understand mesh VoIP QoS characteristics by simulating realistic VoIP conversations over WMNs, with both VoIP and non-VoIP traffic profiles. A sample conversation of a mother and child was translated into a VoIP conversation applying the G.729 codec among 26 mesh nodes. These nodes created 13 VoIP peers. These tests were conducted in three different scenarios and run 20 times. In the no mobility scenario, all of the 26 nodes were stationary, in the limited mobility scenario, 10 out of 26 mesh nodes were mobile and in the full mobility scenario, all the mesh nodes were mobile. Each scenario was tested against two traffic profiles. Profile 1 was VoIP traffic only, using RTP/UDP/IP where no background traffic was present and only nodes were involved in VoIP conversations. Profile 2 was VoIP profile mixed with non-VoIP profile, where non-VoIP profile was simulated by TCP in the greedy mode. The mobile nodes in the limited mobility and full mobility scenarios were configured to move with a walking speed of 1.3m per second. The mobility of the nodes were as per a defined path, where the nodes were starting their movement from their initial positions, moving within the mesh network coverage and finally coming back from where they had started their movements.

5.1 Research conclusion

The conclusion of this research's results is shown in Figure 5-1. It can be summarized as follows: A WMN formed by stationary nodes (no mobility) is an acceptable platform for VoIP implementation, maintaining that there is no background traffic mixed with VoIP traffic. A WMN formed by a mix of stationary and mobile nodes (limited mobility) can be a good platform for VoIP implementation, if there is no background traffic mixed with VoIP traffic. A WMN formed by full mobile nodes is also a good platform for VoIP implementation, if there is no background traffic mixed with VoIP traffic. VoIP implementation with background traffic may not be successful if QoS is not implemented among mesh nodes. The results show that node mobility can result in more packet loss, compared to a scenario in which nodes are not mobile. If node mobility with a speed of 1.3m/sec can increase packet loss, then high mobility of nodes with faster speeds can increase the packet loss to the extent that would make the VoIP implementation unusable. Jitter rate in all the three scenarios didn't cross the acceptable VoIP jitter limit. Therefore a jitter buffer, which is normally used by VoIP applications, can very well solve the problem of jitter in VoIP implementations. The research also shows that if the communicating mobile wireless mesh nodes happen to come close to each other, or are only one hop away, the VoIP conversation can run smoothly even if there is background traffic generated or processed by the communicating nodes. VoIP conversation requires a throughput of an average 40-45 kbps. If this much bandwidth could be allocated to the mesh nodes involved in VoIP conversation, the VoIP traffic throughput requirements of the G.729 codec characteristics and the VoIP traffic profile used herein will be met. The mobile mesh nodes' movement direction has great significance on VoIP QoS. When the mobile nodes are moving towards the direction of their communicating nodes, the VoIP quality improves, but when the mesh nodes are moving against the direction of their communicating nodes, the VoIP quality degrades. Furthermore the movement of mesh nodes serving as the next hop for the VoIP peers, causes increased delay, jitter and packet loss.



	VoIP only	VoIP + non-VoIP	VoIP only	VoIP + non-VoIP	VoIP only	VoIP + non-VoIP
	No mobility		limited Mobility		Full Mobility	
Delay >200ms	0	21	0	23	0	25
Jitter > 100 ms	0	0	0	0	0	0
Packet loss > 5%	0	19	5	22	8	26

Figure 5-1 *Number of nodes exceeding VoIP QoS factor limits*

5.2 Recommendations

The following are some recommendations for VoIP implementation over WMNs, based on the findings of this research. VoIP applications' usage in WMNs is considered to be a cheap solution for rural areas in developing and least developed countries. Urban areas can also make use of WMN to achieve redundancy and high availability for critical services including VoIP services. For a successful VoIP implementation, the QoS issues have to be addressed. Since WMNs currently do not provide QoS capabilities, it is better to create a physically or virtually separated network for VoIP applications, so that other types of traffic won't mix with VoIP traffic. Also, routing protocol choice can help to achieve better QoS. It is recommended that routing protocols should be selected according to the nodes' mobility model. In WMNs design, if the nodes’ moving path could be defined in a way that it moves towards its communication node, the service quality will increase. Furthermore, mesh nodes’ mobility speed should be kept to a minimum, if and when possible.

5.3 Limitations

The following limitations of this research were identified. It was challenging to design the simulation cases using several mobility models; therefore one mobility model was used. Test Cases were run in a simulator, not in a real network. Other limitations include time and hardware constrains to simulate a large WMN of more than 100 nodes and to run test cases using the GOD routing protocol. In this research mesh nodes’ formation were by clients only, no mesh access point and router was used. There was no external interference sources present in the network designs. The average distance between the buildings and the building heights, transmit power, street width and path loss exponent were limited to what has been described in Section 3.2.2. The VoIP profile was scripted to simulate VoIP payload along with RTP/UDP/IP headers. Layer 2 frame header overhead was not considered. The simulation cases used only one type of VoIP profile. VoIP traffic parameterization, considering the common G.729 codec, also proved a challenge. Furthermore, background traffic simulation using TCP in greedy mode was a challenge.

5.4 Future work

WMNs are newly emerged type of networks. Testing various applications over WMNs is necessary to discover if WMNs’ dynamic nature and other unique characteristics result in better or poorer quality of service compared to normal wireless networks. Focusing research efforts on VoIP implementation over WMNs should be of special interest to researchers, due to VoIP’s sensitivity to delay, jitter, packet loss and throughput usage.

WMN implementation in areas where no network infrastructure exists is an ideal choice, but using real time applications over these networks is a challenge. Today people are much more interested in making voice calls instead of video calls, sending e-mails and chatting with friends. In rural areas of countries like Afghanistan and other least developed countries, both basic literacy and computer literacy are big challenges, and this has resulted in widening the digital divide. But today even illiterates can make voice calls, because of its easy-to-use nature. In rural areas WMNs and VoIP applications are also affected by unavailability of reliable electricity. If the WMN equipment is stationary, it needs to be fed with reliable electricity, especially if the equipment serves as WMN access points and gateways. If the WMN equipment is mobile, battery power will have to be used. Since the WMN nodes are allowing other nodes to make use of their resources, the power consumption will be higher, resulting in battery powered equipment quickly going offline., Since the beauty of WMNs lies in its mobility, which can only be achieved by using a battery as the power source; future researches should also address power constraint issues.

Future work based on this research can be focused on any of the following: Testing the same scenarios, but using different WMN routing protocols and analyzing the VoIP QoS factors. Mesh nodes' mobility speed could be modified to a faster speed and its effects on VoIP QoS factors could be studied. Mesh nodes' movement path and direction could be modified to analyze the effects on VoIP QoS factors. The effect of increasing the number of mesh nodes on VoIP QoS, could also be determined by adding more nodes in each scenario. The VoIP profile parameters and characteristics according to other codec types is also a possible field of study. The mesh nodes could be configured to switch to ON and OFF states to study how the WMN topology changes and how the VoIP QoS factors are affected. The influence on VoIP QoS of adding mesh access points and routers as fixed devices could be determined. Using the NCTUns emulation capabilities to communicate with real networks and introducing interferences to degrade mesh coverage, are other possible future studies. Once QoS capabilities are introduced for WMNs, QoS metrics could be configured to prioritize VoIP traffic on each mesh node and to discover how VoIP QoS is affected.

Bibliography

[1] Akunuri, K., Arora, R., & Guardiola, I., G. (2011). A study of speed aware routing for mobile ad hoc networks. *International Journal of Interdisciplinary Telecommunications and Networking, 3*, 40-61.

[2] Akyildiz, I. F., Wang, X., & Wang, W. (2005). Wireless mesh networks: A survey. *Computer Networks, 47*(4), 445-485. www.sciencedirect.com

[3] Anand, K., Samrat, G., Samir, R. D., & Suman, B. (2007). VoIP on wireless meshes: Models, algorithms and evaluation. *INFOCOM 2007. 26th IEEE International Conference on Computer Communications*, 2036-2044.

[4] Bahl, P., Adya, A., Padhye, J., & Wolman, A. (2004). Reconsidering wireless systems with multiple radios. *Computer Communications Review, 34*(5), 39-46. http://portal.acm.org

[5] Bakhshi, A., & Prasanna, V., K. (2004). Energy-efficient communication in multi-channel single-hop sensor networks. *Proceedings of the Tenth International Conference on Parallel and Distributed Systems (ICPADS'04),* 403-410.

[6] Bluetooth Inc. (2007). *Bluetooth.* February/02, 2007, http://www.bluetooth.com/bluetooth/

[7] Bluetooth, I. (2007). *Baseband resource Manager* February/04, 2007, from http://bluetooth.com/Bluetooth/Learn/Works/Core_System_Architecture.htm

[8] Brewer, E., Demmer, M., Du, B., Ho, M., Kam, M., Nedevschi, S., et al. (2005). The case for technology in developing regions. *Published by the IEEE Computer Society,* 25-38.

[9] Caprihan, G., Kumar, S., & Saran, H. (1997). A quality of service translation methodology based on user perception. Proc. *IEEE Singapore International Conference on Networking,* Singapore.

[10] Castro, M., C., Dely, P., Kassler, A., J., & Vaidya, N., H. (2009). QoS-aware channel scheduling for multi-Radio/Multi-channel wireless mesh networks. *Proceedings of the 4th ACM International Workshop on Experimental Evaluation and Characterization,* 11-18.

[11] Chiueh, T., Raniwala, A., Krishnan, R. & Gopalan, K. (2005). *Hyacinth: An IEEE 802.11-based multi-channel wireless mesh network.* February/02, 2007, from http://www.ecsl. cs.sunysb.edu/multichannel/

[12] Choong, K. N., Yee, Y. C., Low, A. L., Chien, S. F., & Ng, S. L. (2006). QoS provisioning for multimedia wireless networks. *BT Technology Journal, 24*(2),117-122.

[13] Cisco Systems. (2006). *Voice over IP - per call bandwidth consumption.* Retrieved 09/22, 2011, from http://www.cisco.com

[14] Cisco Systems. (2011). *What is VoIP?*, 2011, from www.cisco.com

[15] Clausen, T., & Herberg, U. (2011). *Delay tolerant routing with OLSRv2* No. 7662)Institut National De Recherche en Informatique et en Automatique.

[16] Dang, T. D., Sonkoly, B., & Molnár, S. (2004). Fractal analysis and modeling of VoIP traffic. *Telecommunications Network Strategy and Planning Symposium. NETWORKS 2004, 11th International,* , 1-8. doi:10.1109/NETWKS.2004.1341826

[17] Divecha, B., Abraham, A., Grosan, C., & Sanyal, S. (2007). Impact of node mobility on MANET routing protocols models., 1-10.

[18] Dragos, N., Samrat, G., Kyungtae, K., & Rauf, I. (2007). Performance of VoIP in a 802.11 wireless mesh network. Paper presented at the *INFOCOM 2006. 25th IEEE International Conference on Computer Communications,* Barcelona, Spain. pp. 1-11.

[19] Draves, R., Padhye, J., & Zill, B. (2004). Routing in multi-radio, multi-hop wireless mesh networks. *MobiCom 2004,*

[20] El-Kadi, M., Olariu, S., & Abdel-Wahab, H. (2002). A rate-based borrowing scheme for QoS provisioning in multimedia wireless networks. *IEEE Transactions on Parallel and Distributed Systems, 13*(2), 156-166. Retrieved from IEEE Computer Society database.

[21] Galperin, H. (2005). Wireless networks and rural development: Opportunities for latin america. *Information Technologies and International Development, 2*(3), 47-56.

[22] Gambiroza, V., Sadeghi, B., & Knightly, E., W. (2004). End-to-end performance and fairness in multihop wireless backhaul networks. Paper presented at the *The 10th Annual International Conference on Mobile Computing and Networking,* Philadelphia, PA, USA. pp. 287-301. Retrieved from http://portal.acm.org

[23] Ian, A., F. (2005). A survey on wireless mesh network., S23-S30.

[24] Jang, S. (2003). In Kelly B., E., Davis A., W.(Eds.), *A technical FAQ: Frequently asked questions about voice and video over IP networks*. MA, USA: Wainhouse Research. Retrieved from http://www.wainhouse.com/files/papers/wr-faq-ip-conf.pdf

[25] Jian, L., Zhi, L., & Prasant, M. (2008). Adaptive per hop differentiation for end-to-end delay assurance in multihop wireless networks. *Ad Hoc Networks Journal, 7*(6), 1169-1182.

[26] Jun, J., & Sichitiu, M., L. (2008). MRP: Wireless mesh networks routing protocol. *Computer Communications, 31*(7), 1413-1435.

[27] Karam, M., J., & Tobagi, F., A. (2002). Analysis of the delay and jitter of voice traffic over the internet. *INFOCOM 2001. Twentieth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE, 2*, 824-833.

[28] Kaur-Kehal, R., & Sengupta, J. (2011). A comprehensive review on improving QoS for VoIP in wireless mesh networks. *Journal of Global Research in Computer Science, 2*, 32-33.

[29] Kim, H., Yun, S., & Lee, H. (2007). Boosting VoIP capacity of wireless mesh networks through lazy frame aggregation. *IEICE Trans. Commun., E90–B, 5*, 1283-1285.

[30] Kim, K., & Shin, K., G. (2006). On accurate measurement of link quality in Multihop wireless mesh networks. Paper presented at the *MobiCom '06 Proceedings of the 12th Annual International Conference on Mobile Computing and Networking,* pp. 38-49.

[31] Knoblauch, R., L., Pietrucha, M., T., & Nitzburg, M. (1996). Field studies of pedestrian walking speed and start-up time. *Transportation Research Record,* 27-38.

[32] Liese, S., Wu, D., & Mohapatra, P. (2006). Experimental characterization of an 802.11b wireless mesh network. *IWCMC '06 Proceedings of the 2006 International Conference on Wireless Communications and Mobile Computing,* pp. 587-592.

[33] Marwah, K., & Singh, G. (2011). VoIP over WMN: Effect of packet aggregation. *International Journal on Computer Science and Engineering, 3,* 2323-2331.

[34] Mlinarsky, F. (2006). The challenges and importance of testing mesh networks prior to deployment. *Rf Design, 29*(6), 16-23.

[35] Moskaluk, D. (2007). *VoIP using wireless mesh infrastructure.* Retrieved 02/2011, 2011, from http://www.moskaluk.com/voip_using_wireless_mesh_infrast.htm

[36] National Academy of Science. (1994). What is experimental computer science and engineering? [Academic Careers for Experimental Computer Scientists and Engineers] (pp. 9-9). Washington D.C: National Academy of Sciences.

[37] Olivier, S. M. (2004). Experiments. *Information technology research* (Second ed., pp. 68-77). Pretoria, South Africa: Van Schaik.

[38] Poe, R. (2007). *Mesh wi-fi complicates wireless VoIP use.* Retrieved 09/22, 2011, from http://www.focus.com/briefs/mesh-wi-fi-complicates-wireless-voip-use/

[39] Reddy, B., T., John, J., P., & Murthy, S. R., C. (2006). Providing MAC QoS for multimedia traffic in 802.11e based multi-hop ad hoc wireless networks. *Computer Networks*, 154-176. Retrieved from www.sciencedirect.com

[40] Serrano, S., Campobello, G., Leonardi, A., & Palazzo, S. (2011). A MOS-based routing approach for wireless mesh networks. Paper presented at the pp. 1-6.

[41] Sevanto, J., Liljeberq, M., & Raatikainen, K. (1998). Introducing quality-of-service and traffic classes into wireless mobile networks., 21-31. Retrieved from ACM database.

[42] Shao, H. (2004). Multimedia QoS support specified in various wireless network standards. *IEEE MultiMedia, 11*(4), 75-77.

[43] Shuang, Y. (2011). *Extending the base IEEE 802.11™ WLAN standard to include mesh networking*. Retrieved November/07, 2011, http://standards.ieee.org

[44] Strix Systems. (2005). *Solving the wireless mesh network multi-hop delimma* Strix System.

[45] Strix Systems. (2006). *Next steps in municipal wireless mesh networks.* Retrieved 09/22, 2011, from www.wlanmall.com/media/catalog/pdf/strix_muni_mesh.pdf

[46] Tsai, T., & Chen, J. (2005). IEEE 802.11 MAC protocol over wireless mesh networks: Problems and perspectives. *Proceedings of the 19th International Conference on Advanced Information Networking and Applications (AINA'05)*, 60-63.

[47] Vasan, A., & Shankar, U., A. (2002). An empirical characterization of instantaneous throughput in 802.11b WLANs.*CS-TR-4389*, 1-6.Retrieved from http://www.cse.iitb.ac.in

[48] VoIP Foro. (2006a). *Jitter*. Retrieved 2 February 2007,from http://www.en.voipforo.com

[49] VoIP Foro. (2006b). *Latency*. Retrieved 2 February 2007, from http://www.en.voipforo.com

[50] VoIp-info. (2011). *Bandwidth consumption.*, 2011, from http://www.voip-info.org/

[51] Waharte, S., Boutaba, R., Iraqi, Y., & Ishibashi, B. (2006). Routing protocols in wireless mesh networks: Challenges and design considerations. *Multimed Tools Appl*, (29), 285-303.

[52] Wang, S., Chou, C., & Lin, C. (2010). *The GUI user manual for the NCTUns 6.0 network simulator and emulator* Network and System Laboratory, Department of Computer Science, National Chiao Tung University, Taiwan.

[53] Wang, S., & Lin, Y. (2005). NCTUns network simulation and emulation for wireless resource management. *Wireless Communications and Mobile Computing, 5*, 899-916.

[54] Wang, X., Chen, S., & Jajodia, S. (2005). Tracking anonymous peer to peer VoIP calls on the internet. Proc. of *Computers and Communications Security* , pp. 81-91.

[55] Wikipedia Contributors. (2011). *Throughput.* Retrieved 11/23, 2011, from http://en.wikipedia.org

[56] WiMAX Spectrum Owners Alliance. (2007). *Spectrum.* Retrieved February/04, 2007, from http://www.wisoa.net/site/what-is-wimax/spectrum/

Appendix A Unpublished draft paper

An analysis of Voice over Internet Protocol in wireless mesh networks

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Abstract— This paper analyzes the impact upon quality of service for voice over Internet Protocol on wireless mesh networks with mobile nodes and simulates voice traffic on such a mesh network to analyze the following performance metrics: delay, jitter, packet loss and throughput. Wireless mesh networks present interesting characteristics such as multi-hop routing, node mobility, and variable coverage that can impact quality of service. There are three wireless mesh network scenarios each with 26 mesh nodes, a reasonable deployment scenario for a small organizational network for either urban or rural deployment has been considered. In first scenario, all mesh nodes are stationary. In the second scenario, 10 nodes are mobile and 16 nodes are stationary. Finally, in third scenario, all mesh nodes are mobile. The mesh nodes are simulated to move at a walking speed of 1.3m per second. The results show that node mobility can increase packet loss, delay, and jitter. However, the results show that wireless mesh networks can provide acceptable quality of service providing there is little or no background traffic generated by other applications. In particular, the results demonstrate that jitter across all scenarios remains within human-acceptable tolerances. It is therefore recommended that voice over Internet Protocol implementations on wireless mesh networks with background traffic be supported by quality of service standards, otherwise they can lead to service delivery failures. On the other hand, voice-only mesh networks, even with mobile nodes, offer an attractive alternative voice over Internet Protocol platform.

Categories and Subject Descriptors
C.2.1 [**Computer-Communication Networks**]: Network Architecture and Design - *Wireless Communication*

I. INTRODUCTION

This paper presents the analysis of voice over Internet Protocol (VoIP) applications on wireless mesh networks (WMNs). The characteristics of WMNs and VoIP applications cause quality of service (QoS) problems. Studies show that WMNs unique characteristics is achieved by combining the wireless ad-hoc network (MANETs), wireless sensor networks and cellular technologies features[7][16], meanwhile owning these features result in QoS issues for VoIP implementations. VoIP applications characteristics are identified as: sensitivity to delay, jitter, packet loss and use of small packets. Generally the WMNs are considered to be a type of mobile ad-hoc network. The similarities between the two are in multi-hop nature and nodes mobility, but the differences are in the use of gateways, traffic flows, nodes mobility, mobile node role and device energy constrain issues [8] [16].

WMNs usually have dynamic and complex topologies. The next hop can change from time to time and service quality may vary based on the speed of nodes movement, distance from other nodes, obstacles and load on mesh nodes. The IEEE 802.11s standard for WMNs considers the mesh nodes as part of the network infrastructure, while mesh nodes can be stationary or mobile. In this research wireless mesh networks formations will only be by wireless mesh clients. Wireless access points and wireless routers are not considered to be part of the mesh setup in this research. The designs are a combination WMNs and MANETs. Studies show that WMNs introduce more delay, jitter and packet loss and it may be causing problem for the VoIP applications. This research will mainly focus on studying the VoIP applications by discovering how this type of traffic is affected by WMNs. An empirical research by running simulation cases and generating VoIP traffic and non-voice traffic will be conducted. The simulation cases are setup with stationary and mobile nodes.

In order to explain how WMNs affect VoIP implementations, the remainder of this paper is organized as follows: Section II discusses the VoIP QoS related works, Section III presents the research question and methodology, Section IV explains the results and analysis, finally in Section V the research will be concluded.

II. RELATED WORK

A **WMN** is a communications network made up of radio nodes organized in a mesh topology. WMNs often consist of mesh clients, mesh routers and gateways. Another good definition can be: WMNs is a self-healing, self organization and fault tolerant network with dynamic topologies and formed by a mix of/only wireless clients, access points and/or routers. Studies show that real-time traffic in wireless networks requires QoS (QoS) for prioritization [15]. For this reason the IEEE 802.11e MAC (Media Access Control) protocol was proposed to provide QoS. Research done by [14] confirms that since WMNs are characterized as multi-hop transmission, the IEEE 802.11e MAC which deals with QoS does not fit the requirements of backhaul networking in WMNs.

Routing protocols play a key role in order to facilitate mesh nodes discovery and communication. Therefore the routing protocols choice and characteristic can affect the QoS [8]. There are several important factors for choosing a mesh routing protocol like size of network, nodes mobility and type of traffic. Mesh routing protocols are usually classified as proactive, reactive or hybrid. There are many mesh routing protocols, but some of them are commonly used which are listed as follows: B.A.T.M.A.N, DSDV, HSR, IARP, OLSR and DSR [16]. Another routing protocol named GoDRP (God Routing Protocol) is also used by simulation software like ns2 (Network Simulator 2) and NCTUns *(National Chiao Tung University- network simulator)* to calculate the routes based on nodes' position and signal range without any routing protocol overhead, this type of routing protocol is used to help and benchmark the simulations to the best way a routing protocol can theoretically perform [17].

Prior to implementing VoIP applications, it is important to understand and test the existing networks if they can support VoIP applications. Research shows that WMNs characteristics and complexities make it challenging to implement VoIP applications which are mainly due to delay, jitter, packet loss, multi-hop path and dynamic nature [7].

Today VoIP applications are widely used, but still there is lack of QoS for voice applications in the new emerging WMNs, since VoIP applications are exchanging many small packets which are made of big packet headers and small VoIP payloads. Researchers confirm that little efforts has been dedicated to address and investigate these problems on wireless multi-hop networks [9][10]. WMNs QoS is usually affected by delay, packet loss. Usually one-way delay of 200 ms, jitter rate of less than 100ms and packet loss of less than 5% is acceptable in VoIP conversations [2]. Another study by [6] using the common G.729 codec with 20 byte VoIP payload, 50 packets per second shows that taking into account the VoIP silence period can increase the utilization by up to 30%. The silence periods are natural in VoIP conversation where no packets are sent. **VoIP traffic characteristics** are unique, considering their packet size, number of packets per second, inter packet delays and dependencies on the type of codec being used. Studies by [3] [11] explains that on the G.729 VoIP payload can be 10, 20, 30 or 40 bytes, but the default is 20 bytes. Since VoIP uses the RTP (Real-Time Transport Protocol) with a header of 12 bytes, UDP (User Datagram Protocol) with a header of 8 bytes and IP with the header of 20 bytes, in total it makes 40 bytes of RTP/UDP/IP headers. Now if a VoIP payload of 20 bytes is added, it sums up to 60 bytes in total without the consideration of data link headers. G.729 codec with the 20 bytes VoIP payload requires that 50 packets to be sent per second. The number of packets can change if the VoIP payload increases or decreases, but for a normal calculation 50 packets per second is used to study the VoIP traffic. Also VoIP conversation has speech periods and silence periods. Studies show that if VoIP applications use the voice activity detection mechanism, and during the silent periods voice packets are not sent, it saves about 35% of the bandwidth for an average volume of 24 simultaneous calls. Research by [19] shows that the VoIP inter packet delay time is usually between 10 and 30ms. The inter packet delay (IDP) time can even increase when the back-off algorithms senses that medium is busy.

Identifying VoIP flows in real-time is important for researchers in order to manage network traffic issues, prioritize VoIP flows, reserve bandwidth or block calls for some certain destinations. The research by [20] shows that when two persons A and B talk to each other, their conversation can be modeled in four states: A talking, B talking, both A and B talking, both silent. These states can be modeled by Markov 4-state chain. Another study by [4] shows that VoIP conversations are made of talk-spurts (on periods) and silence gaps (off periods) on G.729 codec, since the human conversation also has talk periods and silence periods.

Nodes mobility can affect the performance of mesh routing protocols and QoS. Before a mesh routing protocol is selected the nodes mobility model has to be identified. An empirical study by [5] has compared DSDV with DSR mesh routing protocols. The comparison of these routing protocols was done on four-node mobility models which are Random Waypoint, Random Point Group Mobility, Freeway Mobility and the Manhattan Mobility models. The research results show that DSR performs better than DSDV in high mobility networks, since DSR is having faster route discovery compared to DSDV when the old route is not available.

VoIP critical metrics and factors that affect QoS in WMN are delay, jitter, packet loss and bandwidth [13] & [7]. Besides there are other hidden factors as well like mobility, obstacles and weather conditions that affect the link quality [11]. Mobility is also one of the main factors of measuring mesh QoS [1] &[13], but it is one of most complicated and challenging factors to measure. In order to measure the QoS factors there are different methods and tools used to conduct the tests and do the data collections. Some researchers have developed their own testing and measurement tools and some researchers have used on the shelf tools like Iperf, rude and crude [6], RTP, STG, RTG, STCP, RTCP etc. The next section explains the simulation tools, scenarios and methodology.

III. METHODOLOGY

Referring back to the literature, it was stated that the IEEE 802.11e standard which addresses QoS in the wireless networks is designed for single-hop. Since WMNs are of multi-hop nature, the IEEE 802.11e standard cannot be applied on time. As a result WMNs can be challenging for VoIP implementation by introducing delay, jitter, packet loss and less bandwidth allocation compared to single-hop for VoIP applications [7] &[13]. Therefore this research focuses on the answer for the following question:

How are VoIP QoS factors affected by WMNs node mobility?

The related works show that there hasn't been much work done by other researchers addressing this type of problem in WMNs, especially when the mesh nodes are mobile and stationary. Also the related work shows that QoS for VoIP traffic has still remained as a problem among the research community and more research is required to investigate and understand how WMNs affect VoIP applications. The related works also show that researchers are proposing different solutions, but none of the solutions have solved the problem, this is due to the fact that more research is require in order to study and understand the WMNs affects on VoIP QoS. This research's aim is to investigate and discover the problems that can affect the VoIP implementations. This research will study the VoIP implementation in three different WMNs scenarios. This paper analyzes the QoS critical factors like delay, jitter, packet loss and bandwidth as discussed in related work, and discovers the reasons that affect these factors and why it goes beyond the acceptable limits. This research measures and analyzes all these critical QoS factors for VoIP traffic only, and discovers if VoIP applications can be successful in WMNs considering VoIP and WMNs unique type, nature and characteristics.

Since the focus is on discovering the problems that VoIP applications confront when the mesh nodes are moving and measuring some quantities like delay, jitter, packet loss, and bandwidth an empirical study is required in order to help us discover actual affects of WMNs on service quality[5][12] [15][18]. In this research the wireless mesh nodes have been configured to generate traffic using single-channel multi-hop mesh network. Data collecting, measurements and statistics are based on source and destination nodes. Several QoS factors will be investigated like the amount of delay that is caused by number of hops and load, amount of jitter produced as a result of multiple hops and transmission delay, number of packets lost among the nodes and bandwidth used at the each node by VoIP and non-VoIP applications.

Simulation Software choice for this research is the NCTUns simulation /emulation version 6.0 [18]. This simulation software enables us to design and simulate WMNs. Three WMN topologies have been designed and test cases were run in order to analyze the VoIP applications behavior considering WMNs nodes mobility in each scenario.

Meanwhile data and statistics are collected in order to analyze how VoIP QoS factors are affected by comparing scenarios and traffic profiles. The test cases have been designed in three difference scenarios, each scenario is tested against two VoIP traffic profiles. Profile one is a simple VoIP conversation between two mesh nodes without any background traffic. Profile two is VoIP traffic along with background traffic like TCP greedy.

Experimental design in this research based on a single-channel multi-hop WMN on 802.11b IEEE standard. The 802.11b standard will be deployed among 26 nodes. Using the simulator, three WMNs topologies were designed, each with 26 nodes. All the nodes are operating in ad-hoc mode. The 26 nodes are covering an area of almost 132248 m^2 (y = 488m, x= 271m). Mesh nodes are running the GoD routing protocol [17].

Tx. Status	No. Packets	Inter packet delay in sec		Packet length in bytes		
		Min	Max	Avg	Min	Max
on:	500	0.01	0.05	70	60	80
off:	2					
on:	1500	0.01	0.05	70	60	80
off:	19					
on:	2250	0.01	0.05	70	60	80
off:	19					
on:	3000	0.01	0.05	70	60	80
off:	34					
on:	2250	0.01	0.05	70	60	80
off:	19					
on:	1500	0.01	0.05	70	60	80
off:	19					
on:	2250	0.01	0.05	70	60	80
off:	19					
on:	3000	0.01	0.05	70	60	80
off:	35					
on:	2250	0.01	0.05	70	60	80
off:	19					
on:	250	0.01	0.05	70	60	80
off:	4					

Table 1: *Speaker 1 STG VoIP Configuration Script using silence suppression (Off),Speaker2 configuration script is the same, except the number of packets values are less.*

A **VoIP profile** has been scripted to simulate two persons talking to each other using wireless mesh enabled devices. As a test case a mother talking to her child will be simulated, where the mother does most of the talking. During the VoIP conversation there are occasions when mother (speaker 1, Table 1) and child (speaker 2) are both talking at the same time, mother talking and child listening, child talking and mother listening, or both are silent. When speaker 1 talks, the mesh node is sending VoIP traffic to speaker 2, the traffic will go across the mesh network to reach speaker 2. Mostly, when speaker1talks, speaker2 listens, and vice versa, silence suppression. This model of communication is based on the Markov model [20].

As discussed in section II, VoIP software usually uses the RTP protocol in order to transport voice traffic over UDP and IP. If a VoIP payload size of either 20, 30, 40 bytes is considered and then the RTP/UDP/IP headers are added to VoIP payload it will be make 60, 70 and 80 bytes respectively. In this research the packet sizes of 60, 70 and 80 bytes are considered to simulate the VoIP payload along with the RTP/UDP/IP headers with 50 packets/sec and IPD of 0.01-0.05ms.

In VoIP profile, the mother and child conversation flow can be broken up as follows. Mother starts with short greetings. During the greeting, both mother and child are talking for almost 10 seconds. Here both nodes are talking and generating traffic, therefore both are set to the ON mode. Then they pause for 2 seconds, the OFF mode, and then the mother starts talking for 30 seconds, the ON mode, while the child is listening, the OFF mode. This conversation continues for some time with a sequence of ON and OFF states and then the mother says goodbye to her child and the child responds by a goodbye and the conversation ends.

This conversation takes 562 seconds. In total the mother generates approximately 18250 packets and the child generates 7500 packets for the whole conversation. These numbers of packets are just estimations, and in live applications it can change, since when the voice traffic is packetized it can have varying sizes and this number may increase or decrease depending on the codec being used, here the G.729 codec is considered.

Now in order to simulation a human conversation, a traffic generation tool has to be used that can simulate a human VoIP conversation by generating packets with varying sizes, varying inter packet delays, simulating ON and OFF periods. To achieve this, the STG (sent traffic grapher) and RTG (receive traffic grapher) tools were used The STG tool is used to send traffic and the RTG is used to receive traffic. The STG can be used with several modes like TCP, UDP and configuration which allows us to write a script and translate the human conversation into a form that the STG tool can read from the script to generate the traffic as per the defined VoIP parameters for a human conversation.

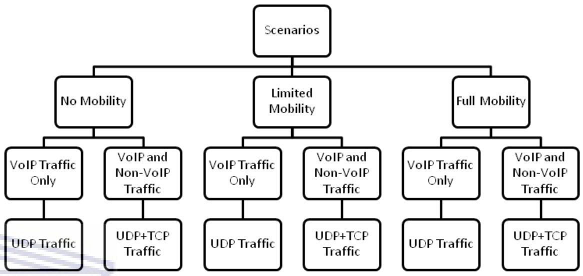

Figure. 1: *Scenarios Design and details, VoIP(UDP) and non-VoIP(TCP) profile summaries*

The **non-VoIP profile** background traffic is simulated using the STCP and RTCP traffic generation tools. Here a simple TCP greedy traffic mode, where the tool establishes numerous TCP connections between the two communicating nodes and transmits TCP data. This traffic is generated to simulate the background traffic. The test cases of VoIP only and VoIP with non-VoIP traffic profiles are simulated in three different scenarios as shown in Figure. 1. Each one of the 26 nodes is configured with following traffic generation tool commands. As an example, Node1 and Node 25 configuration commands are explained for profile 1 and profile 2. All other nodes are configured the same way in all scenarios.

Node 1 configuration for VoIP only profile.
stg -i spkrconfig1.cfg -p 4000 1.0.1.25
rtg -u -p 4000 -w pktlog1 -o thrlog1
Node 25 Configuration for VoIP only Profile:
stg -i spkrconfig2.cfg -p 4000 1.0.1.1
rtg -u -p 4000 -w pktlog25 -o thrlog25
Node 1 configuration for VoIP and non-VoIP Profile:
stg -i spkrconfig1.cfg -p 4000 1.0.1.25
rtg -u -p 4000 -w pktlog1 -o thrlog1
rtcp -p 5000 -w rtcplog1
Node 25 Configuration for VoIP and non-VoIP Profile:
stg -i spkrconfig2.cfg -p 4000 1.0.1.1
rtg -u -p 4000 -w pktlog25 -o thrlog25
stcp -p 5000 1.0.1.1

No mobility, limited mobility and full mobility scenarios are formed by 26 mesh nodes. In *No mobility scenario* all the 26 nodes are stationary and they don't have any movement. Each node is involved in a VoIP conversation with another node. So there are 13 mesh-node peers communicating to each other, shown in Table 2.

Spkr 1	N 1	N 2	N 3	N 4	N 5	N 6	N 7	N 8	N 9	N 10	N 11	N 12	N 13
Spkr 2	N 25	N 15	N 26	N 14	N 22	N 21	N 18	N 16	N 17	N 24	N 20	N 19	N 23

Table 2: *Speaker 1 & Speaker 2 VoIP peers information for all scenarios. In total 13 VoIP peers are communicating.*

Limited mobility scenario is designed and configured with 10 nodes moving at a walking speed of 1.3m/sec, while the other 16 nodes are stationary. All moving nodes travel along the pre-defined path and return back to their original positions. The aim of this scenario is to discover the affects of mobile and stationary nodes on the VoIP traffic. *Full mobility scenario* is designed and configured to simulate all the mesh nodes moving in a walking speed of 1.3m/sec, shown in Figure 2. All nodes move to a pre-defined path (gray lines) and come back to their original position.

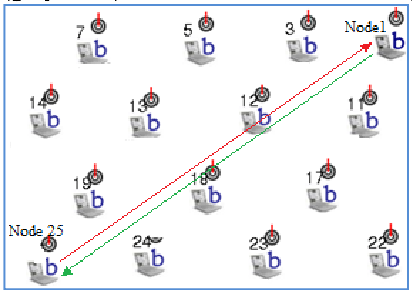

Figure. 2: *All 3 scenarios have similar screenshots, in no mobility scenario all nodes are stationary, in limited mobility scenario nodes: 2, 5, 8, 9,11, 13, 16, 21, 23 & 25 are mobile and in full mobility scenario all nodes are mobile.*

V. RESULTS

Analyzing each one of the VoIP QoS factors, Figure.3 shows that delay in VoIP only profiles are the lowest, regardless of the node mobility factor. VoIP traffic delay in scenarios with background traffic is mostly higher than 200ms, only in a few nodes where the VoIP peers are only one hop away, the delay is lower than 200ms.

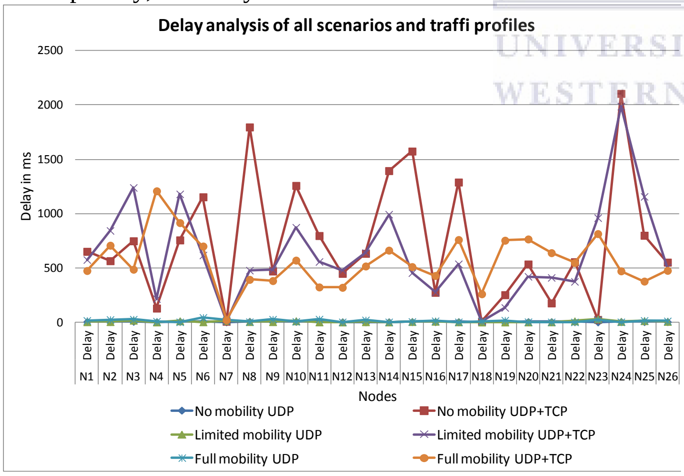

Figure. 3: *In all scenarios VoIP only profiles have less delay compared to profiles with background traffic.*

Jitter analysis (Figure. 4) shows that scenarios with VoIP only profiles have lower jitter rates, while scenarios with background profile have higher jitter rates. In all scenarios the jitter limit of less than 100ms is not crossed.

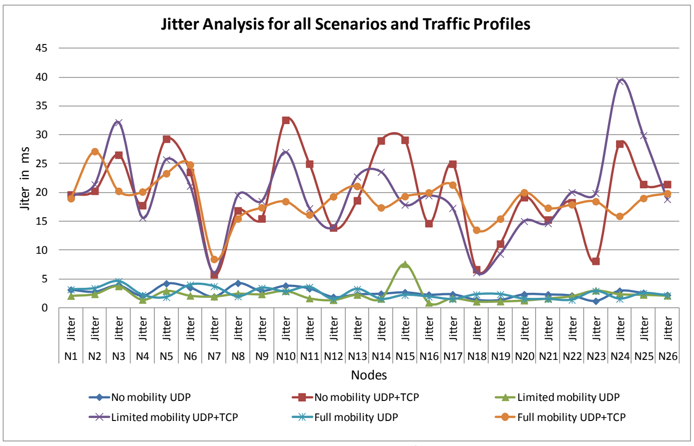

Figure. 4: *In all scenarios VoIP only profiles have lower jitter rate compared to VoIP profiles with background traffic.*

The packet loss analysis (Figure 5) shows that nodes mobility increases packet loss. Even scenarios with VoIP only profiles have packet loss rates above 5%. Scenarios with background traffic have packet loss reaching up to 80% rate. Only in a few nodes, where VoIP peers are one hop away, does it fall below 5% rate.

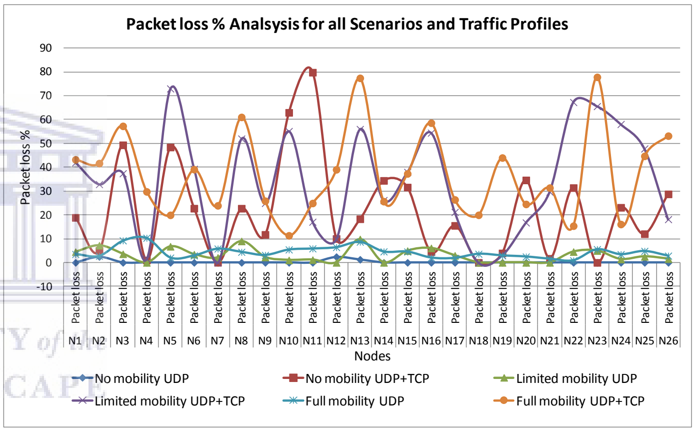

Figure. 5: *Nodes Mobility increases packet loss. All scenarios have lower packet loss rate in VoIP only profiles.*

Throughput analysis (Figure. 6) shows that scenarios with VoIP only profile require 40-45kbps bandwidth. In scenarios with background traffic, bandwidth allocation and usage is still on the same range, but traffic prioritization and use of priority queues are required, since both types of traffics are using the normal queues.

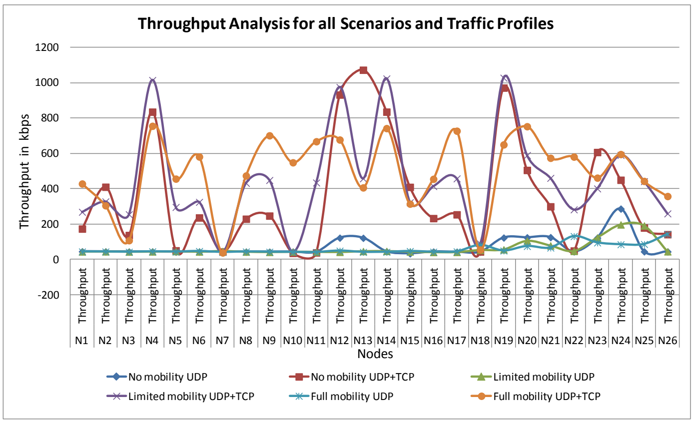

Figure. 6: *All scenarios, VoIP only profiles required 40-45 kbps*

VI. CONCLUSION AND FUTURE WORK

VoIP applications in WMNs, whether nodes are stationary or mobile, can be successful if no background traffic is mixed with VoIP traffic. VoIP implementation with background traffic may not be successful if QoS is not implemented among mesh nodes; Nodes mobility can result more packet loss; If nodes mobility with a speed of 1.3m/sec causes packet loss, then high node mobility can increase the packet loss to an extend that would make the VoIP implementation unusable; Jitter rate in all the three scenarios didn't cross the acceptable VoIP jitter limit, therefore a jitter buffer can very well solve the problem of the jitter in VoIP implementations; If the VoIP enabled wireless mesh nodes happen to be close to each other or only one hop away, the VoIP conversation can run smoothly even if there is background traffic generated or processed by the VoIP enabled nodes; VoIP conversation requires a throughput of an average 40-45 kbps; The mobile mesh nodes movement direction has great significance on VoIP traffic quality. If a mobile node is moving towards the direction of its communicating node, the VoIP quality improves, if the mesh node is moving against the direction of its communicating node the VoIP quality degrades.

Recommendations for VoIP implementation over WMNs are as follows: Until QoS standards are not supported by WMNs, it is better to create a separate network for VoIP applications, so that other traffic won't mix with VoIP traffic; Proper routing protocol according to the nodes' mobility model has to be selected; In WMNs design, nodes moving path should be defined in such a way that it can move towards its communicating nodes, where possible; If and when possible mesh nodes mobility speed should b kept to minimum.

Limitations of this research is as follows: Simulating one mobility model; Running test cases in a simulator, not in a real network; Time and hardware constrains in order to simulate a large WMN of more than 100 nodes; Simulating test cases using GoD routing protocol; Mesh nodes formation by clients only; Usage of one type of VoIP profile; Background traffic simulation using TCP greedy; VoIP traffic parameterization considering the common G.729 codec.

Future work based on this research can be focused on any of the following; Testing the same scenarios, but using different WMN routing protocols and analyzing the VoIP QoS factors; Modifying mesh nodes mobility speed to a faster speed and study its affects on VoIP QoS factors; Modifying mesh nodes movement path and direction and then analyzing its affects on VoIP QoS factors; Increasing the number of mesh nodes and studying its affects on VoIP QoS factors; Changing the VoIP profile parameters and characteristics according to other codec types and studying its affects on VoIP QoS factors; Configuring the mesh nodes to switch to ON and OFF states and then studying how the WMN topology changes and how the VoIP QoS factors are affected; Adding mesh access points and routers as fixed devices and studying their affects on VoIP QoS.

Note: This is an unpublished draft paper.

REFERENCES

[1] Akunuri, K., Arora, R., & Guardiola, I., G. (2011). A study of speed aware routing for mobile ad hoc networks. *International Journal of Interdisciplinary Telecommunications and Networking, 3*, 40-61.

[2] Anand, K., Samrat, G., Samir, R. D., & Suman, B. (2007). VoIP on wireless meshes: Models, algorithms and evaluation. *INFOCOM 2007. 26th IEEE International Conference on Computer Communications*, 2036-2044.

[3] Cisco Systems. (2006). *Voice over IP - per call bandwidth consumption.* http://www.cisco.com

[4] Dang, T. D., Sonkoly, B., & Molnár, S. (2004). Fractal analysis and modeling of VoIP traffic. *Telecommunications Network Strategy and Planning Symposium. NETWORKS 2004, 11th International,* 1-8.

[5] Divecha, B., Abraham, A., Grosan, C., & Sanyal, S. (2007). Impact of node mobility on MANET routing protocols models. *SoftComputing.net*, 1-10

[6] Dragos, N., Samrat, G., Kyungtae, K., & Rauf, I. (2007). Performance of VoIP in a 802.11 WMN. Proc. *INFOCOM 2006. 25th IEEE International Conference on Computer Communications,* Barcelona, Spain. 1-11

[7] Jian, L., Zhi, L., & Prasant, M. (2008). Adaptive per hop differentiation for end-to-end delay assurance in multihop wireless networks. *Ad Hoc Networks Journal,*

[8] Jun, J., & Sichitiu, M., L. (2008). MRP: Wireless mesh networks routing protocol. *Computer Communications Butterworth-Heinemann, 31(7),* 1413-1435

[9] Kaur-Kehal, R., & Sengupta, J. (2011). A comprehensive review on improving QoS for VoIP in wireless mesh networks. *Journal of Global Research in Computer Science, 2*, 32-33.

[10] Kim, H., Yun, S., & Lee, H. (2007). Boosting VoIP capacity of wireless mesh networks through lazy frame aggregation. *IEICE Trans. Commun., E90–B, 5*, 1283-1285.

[11] Kim, K., & Shin, K., G. (2006). On accurate measurement of link quality in Multihop wireless mesh networks. Proc. *MobiCom '06 Proceedings of the 12th Annual International Conference on Mobile Computing and Networking*, 38-49.

[12] Liese, S., Wu, D., & Mohapatra, P. (2006). Experimental characterization of an 802.11b wireless mesh network. Proc. *IWCMC '06 Proceedings of the 2006 International Conference on Wireless Communications and Mobile Computing*, 587-592.

[13] Mlinarsky, F. (2006).The challenges and importance of testing mesh networks prior to deployment. *Rf Design, 29*(6),16-23.

[14] Reddy, B., T., John, J., P., & Murthy, S. R., C. (2006). Providing MAC QoS for multimedia traffic in 802.11e based multi-hop ad hoc wireless networks. *Computer Networks*, 154-176.

[15] Shao, H. (2004). Multimedia QoS support specified in various wireless network standards. IEEE MultiMedia, 11(4), 75-77.

[16] Waharte, S., Boutaba, R., Iraqi, Y., & Ishibashi, B. (2006). Routing protocols in wireless mesh networks: Challenges and design considerations. *Multimed Tools Appl*, (29), 285-303.

[17] Wang, S., Chou, C., & Lin, C. (2010). *The GUI user manual for the NCTUns 6.0 network simulator and emulator* Network and System Laboratory, Department of Computer Science, National Chiao Tung University, Taiwan. http://*nsl10.csie. nctu.edu.tw*

[18] Wang, S., & Lin, Y. (2005). NCTUns network simulation and emulation for wireless resource management. *Wireless Communications and Mobile Computing, 5,* 899-916.

[19] Wang, X., Chen, S., & Jajodia, S. (2005). Tracking anonymous peer to peer VoIP calls on the internet. Proc. *Computers and Communications Security,* 81-91.

[20] Wu, C., Chen, K., Chang, Y., & Lei, C. (2008). Detecting VoIP traffic based on human conversation patterns. *Systems and Applications of IP Telecommunications*, 1-8