Performance Evaluation of AODV and DSR Routing Protocols with PCM and GSM Voice Encoding Schemes

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ABSTRACT

A mobile ad hoc network (MANET) is one of the narrowest and most specific of research topics in the field of telecommunications**.** The growth of this type of network, and the large number of applications with mobility requirements, has led to a wider study and research in the analysis and enhancement of the work in this area. In such networks, nodes are communicating with each other without the need of a centralized administration (This type of network does not contain any type of server or base station). In this topology, the communication between the nodes is done by pair to pair within the coverage area. The routing is managed and organized by a number of routing protocols. A limited coverage area, collision and power consumption for mobile nodes are the main problems occurring in such networks.

In this thesis, two important MANET routing protocols were used, AODV and DSR to analyze their behavior with two different voice encoding schemes, Pulse Code Modulation (PCM) and Global System Mobile (GSM). The PCM and the GSM encoding voice schemes are evaluated with a different number of clients using a Random Way Point Mobility (RWPM) model. OPNET simulator version 17.1 was used to build the modeler and to simulate the ad hoc mobile network model. The benefit of this simulation program is the ability to build models for different network topologies and the large number of available choices for node performance statistics. In addition to that, results are more confident and accurate compared to other simulation programs found in the literature.

From the analysis of the simulations, it was concluded that, in all cases, the AODV protocol performed better than the DSR protocol. This is because AODV doesn't save the entire possible path from source to the destination node. It takes the newest and most refreshable one. On the other hand, DSR caches all possible paths to the destination. It is also shown that PCM performance is better and more quality than GSM in most of the performance metrics except end -to- end delay, for both AODV and DSR routing protocols.

Keywords: OPNET simulator, Mobile Wireless Ad Hoc Network, AODV, DSR, PCM, GSM.

ÖZ

Mobil özel amaca yönelik ağ (MANET) telekomünikasyon alanında en dar ve özel araştırma konularından bir tanesidir. Son yıllarda, bu tür ağların kullanımında ve hareket gerektiren uygulamaların sayısında artış yaşanmıştır. Bu durum da ilgili alanda daha geniş çalışma ve araştırmaların yapılmasına yol açmıştır. Bu ağlarda, düğümler (node) merkezi bir yönetime ihtiyaç duymadan birbirleri ile iletişim kurabilmektedirler. Bu tür ağlar, herhangi bir sunucu ya da baz istasyonu içermemektedir. Bu topolojide, iletişim, kapsama alanı içindeki düğümler tarafından çiftli yapılmaktadır. Yönlendirme, bir dizi yönlendirme protokolleri tarafından yönetilmekte ve organize edilmektedir. Mobil düğümler için, sınırlı bir kapsama alanı, çarpışma ve güç tüketimi bu tür ağlarda meydana gelen başlıca sorunlardır.

Bu tez çalışmasında, iki önemli MANET yönlendirme protokolü olan AODV ve DSR'nin iki ayrı ses kodlama şeması Darbe Kod Modülasyonu (PCM) ve Küresel Sistem Mobil (GSM)'deki davranışları analiz edilmeye çalışılmıştır. PCM ve GSM kodlama ses şemaları, farklı istemci(client) sayıları ile bir Random Way Point Mobility (RWPM) model kullanılarak değerlendirilmiştir. OPNET simülatörü 17.1 sürümü modelleyici inşa etmek ve ad hoc mobil ağ modelini simüle etmek için kullanılmıştır. Bu simülasyon programının yararı, farklı ağ topolojileri ve çok sayıda kullanılabilir mevcut düğüm performans istatistikleri için, modelleri inşa etme yeteneğinin olmasıdır. Buna ek olarak, sonuçlar literatürde bulunan diğer simülasyon programları ile karşılaştırıldığında daha güvenli ve doğru olduğu tespit edilmiştir.

Gerçekleştirilen simülasyonları analizlerinden, AODV protokolünün her durumda, DSR protokolden daha iyi performans sağladığı sonucuna varılmıştır. Çünkü AODV kaynaktan hedefe, gidilecek düğüme ulaşmak için olası yolların tümünü kaydetmemektedir. AODV, olabilecek en yeni ve en yenilenebilir yolu almaktadır. Diğer taraftan DSR, hedefe ulaşmak için mümkün olan tüm yolları önbelleğine almakta, buda daha fazla çarpışmaya ve gecikmeye neden olmaktadır. Ayrıca bu çalışma, PCM performansının, performans ölçümlerinin çoğunda, uçtan uca gecikme hariç, AODV ve DSR yönlendirme protokollerinin her ikisi içinde daha iyi ve daha kaliteli olduğunu göstermiştir.

Anahtar Kelimeler: OPNET simulatörü, Mobil kablosuz özel amaca yönelik ağlar, AODV, DSR, PCM, GSM

DEDICATION

To My Mother

ACKNOWLEDGMENT

I would like to thank Asst. Prof. Dr. Gürcü Öz.

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LIST OF ABBREVIATIONS

AODV	Ad hoc On demand Distance Vector
CTS	Clear To Sent
DSR	Dynamic Source Routing Protocol
ESS	Extended Service Set
GSM	Global System Mobile
IBSS	Independent Base Service Set
IBSS	Infrastructure Basics Services Set
GSM	Global System Mobile
MANET	Mobil Ad hoc Network
OPNET	Optimizatized Network Engineering Tool
PCM	Pulse Code Modulation
QoS	Quality of Services
RERR	Route Error Packet
RREP	Route Replay Packet
RREQ	Route Request Packet
SIP	Session Initiation Protocol
VoIP	Voice Over IP
WLAN	Wireless Local Area Network

Chapter 1

INTRODUCTION

Wireless networks are becoming more and more important in our life. These networks have many applications in different fields. Studies have been done conducted on how best to enhance the services of the applications installed in the devices of these networks [1].

Some of the most accomplished studies on the hardware parts is more connected to more bandwidth provision, power saving, high bit rate and low delay. However the work on, the software aspects has witnessed new developments in providing good services and security for the programs installed in these wireless devices.

Mobile ad hoc network is a new form of independent network. It is a set of wireless mobile nodes and works without centralized administration. MANET is suitable for applications that require mobility. However, there are also a number of problems in these networks. Restriction of coverage area and nodes mobility sometimes causes problems in communications. Moreover, it is necessary to increase the security and authentication of programs installed on these nodes.

Quality of Service (QoS) is a very important point that should be considered in the evaluation of the performance of these networks. Providing best QoS for MANET networks is one of the main aims of this research work.

In this thesis, two important MANET protocols were analyzed: AODV and DSR. In addition, these routing protocols were used on mobile node with two encoding schemes PCM (Pulse Code Modulation) and GSM (Global System Mobile). The behavior of the AODV and DSR with these voice encoding schemes with a different numbers of clients and nodes was studied. OPNET 17.1 was used for modeling a mobile ad hoc network. The OPNET modeler is one of the best simulation programs. It is used to design and analyze networks of different network topologies. The OPNET modeler provides very accurate and reliable results.

The following bellow is related work for this thesis:

In [2] a simulation with a MANET network is done. AODV and DSR ad hoc mobile routing protocols with simulation time 900sec and data rate 11Mpbs settings are used. The GSM voice application was set for 20, 40 random waypoint mobility nodes movement with 500m random waypoint mobile area. The simulation area was set 5Km ×5Km. Network load, route discovery time, packet end-to-end delay and number of hops per route performance metrics were used to evaluate the MANET work. According to the results AODV protocol is better compared with DSR protocol.

In [3] AODV, DSR, GRP, OLSR and TORA protocols were used with simulation time 300 sec and 600 sec for 25, 85 number of mobile nodes. PCM and GSM voice encoding schemes used with random way point speed 5 and 10 m/sec for simulation area 800m×800m and 1600 m×160m. Performance metrics used are jitter, throughput, end -

to-end delay variation, WLAN delay variation and packet delay variation. From the results it is observed that, AODV is better than other MANET routing protocols.

In [4] performance metrics like throughput and delay are analyzed for 50, 70 and 100 mobile nodes for DSR MANET routing protocol. It is observed that throughput is more in 100 nodes than 50 and 70. Also the delay is less for 70 nodes than 50 nodes.

In [5] it is shown that the link breakage is more in large node density. In the case of DSR, when the route breakage in a network with a large density node occurs, packets are cached and a route repair takes place. This improves the overall throughput of the system.

In [6] OLSR and AODV and DSR protocols were used. It is shown that when the number of nodes increases to 50, the DSR performance is better.

In [7] the performance of AODV, OLSR, GRP and DSR are evaluated. It is shown that, DSR does not perform well for media access delay and WLAN throughput and network load with 60 minutes simulation time for 20 nodes. The mobility node speed is 7m/sec. PCM voice encoding schemes were used. AODV is better than the others routing protocols. In addition, AODV and OLSR perform better compared with DSR.

In [8] there is a simulation work for MANET with AODV, DSDV, TORA and DSR routing protocols with simulation time 900sec and number of nodes 21.TheVoice application scheme is (G.729). The simulation area is 340m × 340m with random waypoint mobility mode. Mobility speeds are (1, 5, 10, 15, 20) meters/sec. Voice packet delay and Packet delivery fraction are evaluated in MANET. DSR is worse compared with other MANET routing protocols.

It is indicated that increasing the number of intermediate nodes resulted in an increase in the number of alternative routes to the destination. This is because the increase in the number of nodes also increases the possibility of finding a destination. Additionally, the AODV performed better under the low mobility nodes. As the number of nodes increased, the nodes behaving as intermediate nodes also increased so the neighbor discovering time was minimized. Furthermore, the route discovery time in DSR also decreased. This is due to the large number of alternative multiple routes to the destination node which is cached in its memory during the route discovery time. In DSR, nodes can store multiple routes in their route cache, the source node can check its route cache for a valid route before an initiating route discovery, and if a valid route is found, there is no need for a route discovery. In this case, since, the routes stored in the route cache will be valid longer; it is more beneficial in a low mobility network. Table 1.1 below shows the summary of related works

In this study AODV and DSR routing protocols were used. Simulation area is 5km×5km with random waypoint mobility area 500m. PCM and GSM voice schemes were used with physical standard 802.11b and data rate up to 11Mbps.

Two groups of performance metrics are used here. The first group is for MANET routing protocols and the second one for voice packets.

Table 1.1: Summary of related work

Ref	Simulation program	Routing protocol	Simulation time	No. of nodes	Application, 802.11 Version	Mobility Model, Movement area,Node speed	Performance metrics	Network size
[2]	OPNET 14.5	AODV, DSR	900 sec	20, 40	GSM , 802.11b	RWAP and 500 m ,-	Network load, Route discovery time, Packet end-to-end delay, No of hopes per route	5 km× 5km
[3]	OPNET 14.5	AODV, DSR GRP, OLSR, TORA	300, 600 sec	25, 85	PCM(G.711) GSM(GSM-EFR), -	RWAP, - ,5 and 10msec	voice Mos, jitter, end –to-end delay variation, wlan delay variation, packet delay variation	800m× 800m, 1600m× 1600 m
[7]	OPNET 14.5	AODV, DSR OLSR, GRP	60 minutes	20	PCM (G.711) TCP(high load)	RWAP, -, 7m/sec	Throughput, Media access, delay, load	-
[8]	NS-2	AODV, DSDV, TORA, DSR	900 sec	21	G.729 Voice encoding schemes.	-, 1, 5, 10, 15, 20 m/sec, -	E2E delay, Packet delivery ratio	340m× 340 m

In this thesis, more information about the MANET with voice application is identified. In Chapter 2, the different models of IEEE 802.11 standard is explained. An overview about the versions 802.11 standard is provided. In Chapter 3, the ad hoc network is described, the basic routing types are studied and detailed information about the two mobile ad hoc routing protocols AODV and DSR is presented. In Chapter 4, voice over IP (VoIP) and the two voice application encoding schemes PCM and GSM are explained. In Chapter 5 the steps used in modeling and building simulation for mobile ad hoc network with the OPNET program are demonstrated. In chapter 6, results were provided and discussed. Finally, in Chapter 7, conclusions are drawn about the work of the MANET with voice application.

Chapter 2

WIRELESS LOCAL AREA NETWORK (WLAN)

IEEE802.11 STANDARD

2.1 Introduction

WLAN is a connection between two or more devices using a spread-spectrum distribution model. The access point provides the connection to the internet for WLAN devices. WLAN is based on IEEE 802.11 standard and carries the name of wi-fi.

F. R. G. Feler and U. Bap conducted research for a WLAN network using diffused infrared connection [1] then; P. Ferert undertook a study for a single code spectrum for WLAN devices connection. [1] K. Pahlvan in IEEE Computer made a comparison between an infrared and a CDMA model connection [1]. The IEEE 802.11 standards were developed with different versions.

There are advantages and disadvantages in wireless LANs networks. One of the most significant advantages is that WLAN has a flexible mode. Movement within the workplace becomes easier and it is less costly to install, unlike a wired mode which is more expensive in terms of time and money. One of the disadvantages of WLAN is that it provides less security and less data rate for transmission.

However, recent developments in the 802.11 standard, have speeded up its effectiveness with more data rate transmission and a higher quality of service.

2.2 Structure of WLAN

There are three types of wireless local area network topologies that can be built from a number of connecting stations, which support an IEEE 802.11 standard. One of them has a control model shape (dependent stations). A second one has no control model shape (independent stations). The third type is a combination of the dependent and independent model shape.

The first model is the Basic Service Set (BSS) which is a logical group of 802.11 stations that require a special device to provide control for data transmission between the stations (permission, rejection). This device is called access points. The client stations don't communicate with each other directly. They communicate with each other by sending a frame from the source to the destination across the access point. Then the access point controls the traffic by forwarding the frames to the destination.

The access point may be wired connected to another network, in which case the model is called Infrastructure BSS. This type of topology represents a bridge between the wireless and the wired network environment. Infrastructure BSS is provides more security and authentication for authorized transmission of encrypted data between stations as in Figure 2.1 shows BSS [9-10].

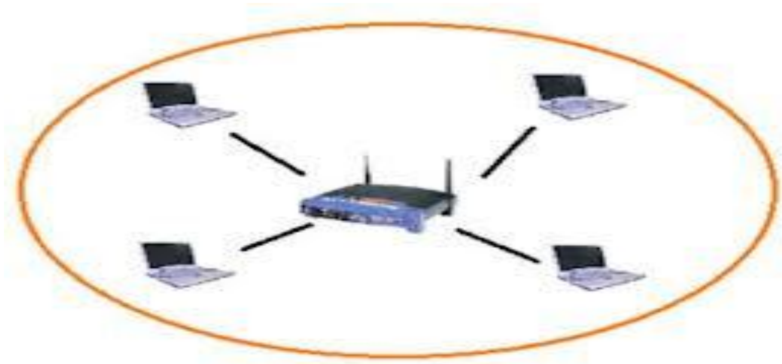


Figure 2.1.Basics Services Set (BSS) [10]

The second model is the Independent Basic Service Set (IBSS), which is a logical set of device stations connected to each other by broadcasting signals. An example of this model is the ad hoc network. There is no controller to the stations' access point. One block of IBSS contains at least two devices. IBSS is used for short time service: e.g., short media conversation [2-3], as shown in Figure 2.2.

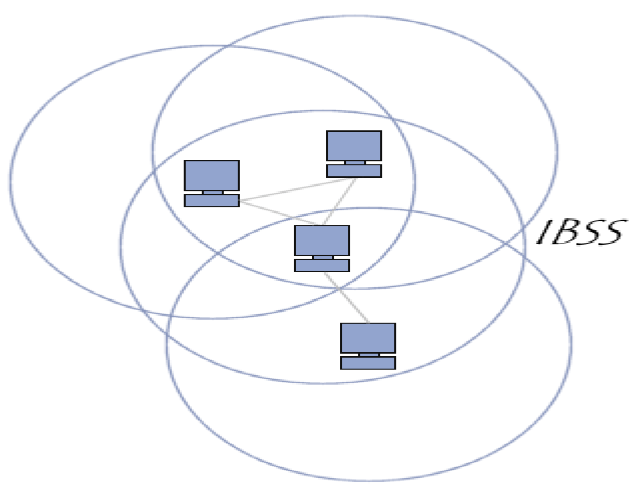


Figure 2.2 Independent Base Service Set (IBSS) [10]

The third model is the Extended Service Set (ESS) which is comprised of several BSS blocks. Each group communicates with other networks across a Distributed Systems

(DS) link. ESS is created by merging two or more different BSS blocks to increase the coverage area. Figure 2.3 shows the ESS network type [11].

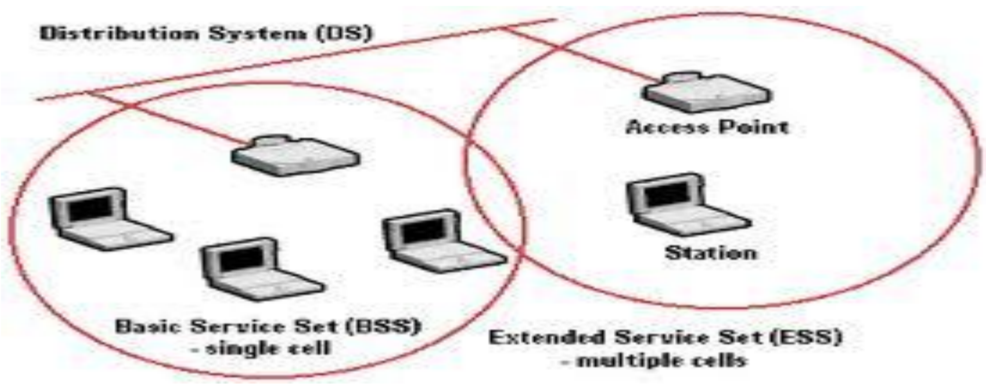


Figure 2.3 Extended Service Set (ESS) [11]

2.3 IEEE 802.11 WLAN Standard

802.11 standard was developed in 1997 by IEEE (Institute Electrical and Electronics Engineers). Since then, more development has been done for versions of IEEE 802.11 by increasing the rate of data transmission and the number of services that can be provided. IEEE802.11 standard is occupying the two lower layers of the ISO model (data link layer and physical layer) as shown in Figure 2.4.

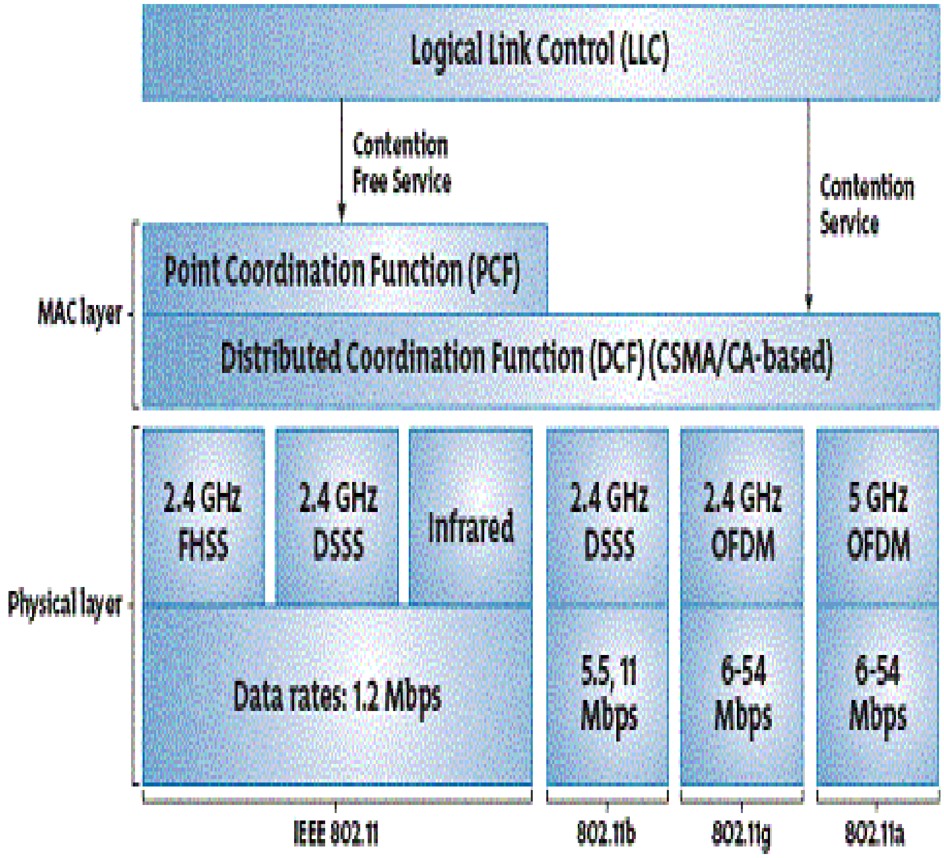


Figure 2.4 The Architecture Of Data Link and physical layer [12]

The logical link sub-layer is a part of the data link layer in the OSI model within IEEE 802.11 standard. It represents an interface in the upper layer that provides control and coordination to the data transmission within the different stations.

Media Access Control (MAC) sub layer provides a reliable data transmission over a wireless environment. It provides a management of the shared medium between the stations and organization of the transmission of data by using Carrier Sense Medium Access with Collision Avoidance (CSMA/CA).

Distributed Coordination Function, provides some kind of management for shared medium between different stations. In addition, mobile stations have a problem of hidden nodes. This problem is solved by using a permission to speak mechanism. This can be done by using three types of the frame controls (RTS, CTS and ACK). Figure 2.5 shows the hidden node problem.

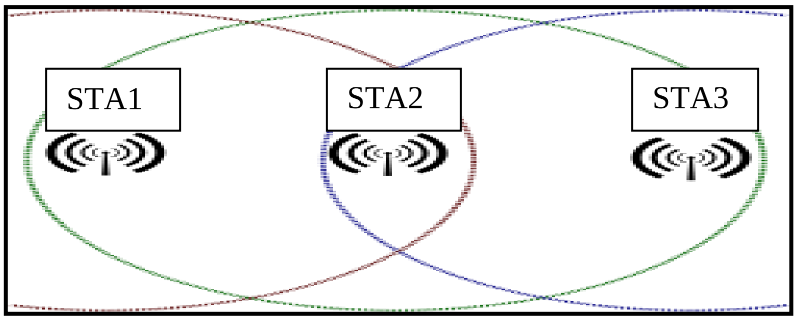


Figure 2.5 Hidden Node Problem

Request to Send (RTS) and Clear to Send Frame Control (CTS) was used to reduce frame collisions caused by the hidden node issue.

When a source node wants to send data to a destination node, the source node sends a control frame (RTS) to a destination node. The destination node replies with a control frame (CTS) to the source node. These control frames were designed for solving collisions caused by the hidden node problem. These protocols were designed on the assumption that all the network nodes were in the same transmission range. Figure 2.6 presents a generation of RTS and CTS between source and destination nodes.

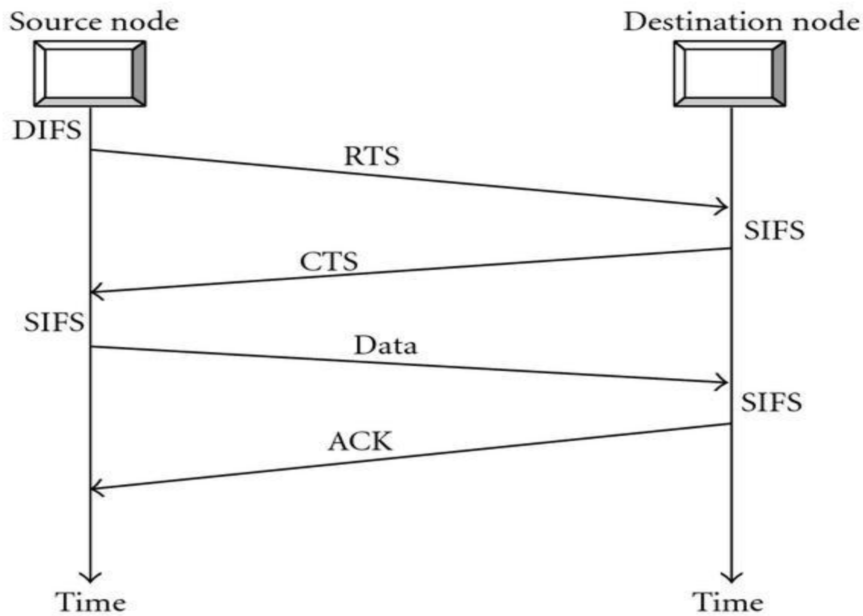


Figure 2.6 RTS/CTS Operation [13]

2.4 Physical Layer and Versions of IEEE 802.11 Standard

Physical layer in IEEE 802.11 provides an interface between the MAC layer and the air interface. It also provides management and control of the frames exchanging between the MAC layer and the physical layer. In addition, the physical layer limits the noise. There are four physical characters, defined by the IEEE standard:

- The first type is Direct Sequence Spread Spectrum (DSSS).

- The second type is the frequency hopping spread spectrum (FHSS), with a different number of channels depending on the country, for example, in Japan twenty three channels are available while in the United States of America it is seventeen.

- The third type is Infrared (IR). This type of the model has become very popular on the market.

- The fourth type is Orthogonal Frequency Division Multiplexing (OFDM). IEEE 802.11 standard has a number of versions.

The first one is the IEEE 802.11a. This version of IEEE 802.11 standard was developed in 1999 and was on the market in 2000. It was developed to work with multi carrier modeling using different frequency multi carrier signals with the OFDM using (5 GHz) frequency bands. IEEE802.11a has a data rate for transmission up to 54 Mbps. This standard offers more non overloading channels up to 12 channels making it more suitable for large data transmission rates in areas like, multimedia, streaming voice and video etc [13].

The second version is the IEEE 802.11b standard which has a bandwidth reaching 11Mbit/s using frequency band 2.4GHz. This version is used in WI-FI communication systems [14].

The third version is the IEEE802.11g. The type of modular, for this version, is OFDM and DSSS with a frequency band up to 2.4 GHZ and data rate up to 54 Mbps [13-14].

Chapter 3

OVERVIEW OF AD HOC ROUTING PROTOCOLS

3.1 Introduction

An ad hoc network is a grouping of wireless mobile devices designed without any centralized administration. The node in this type of network, was acting as a router / host to routing data packet through the network. These transmission stations require a high data rate transmission for voice and video.

3.2 Types of Routing

For a better understanding of the mobile ad hoc network (MANET), we should take a look at the commonly known routing protocol types and also understand the way they work. There are three types of routing protocols; flooding routing, link state routing and source routing.

In flooding routing, there is no need to save any path information about the packet transmission between the source and the destination. The packet is sent to all possible nodes in the network except the link from which the packet came from. As a result of this, more than one copy of the packets will arrive at the destination. Each packet is assigned a unique number to avoid any duplication. Duplicated packets are immediately ignored, which limits the network load as nodes are able to remember the packet from which it has already been forwarded. The Figure 3.1 below represents the three steps in the flooding routing. There is no duplicated transmission of the packet. The packet is sent to all the possible routes on its way. Each source has a different number of hops to reach the destination. This method is very useful for broadcasting transmission [15].

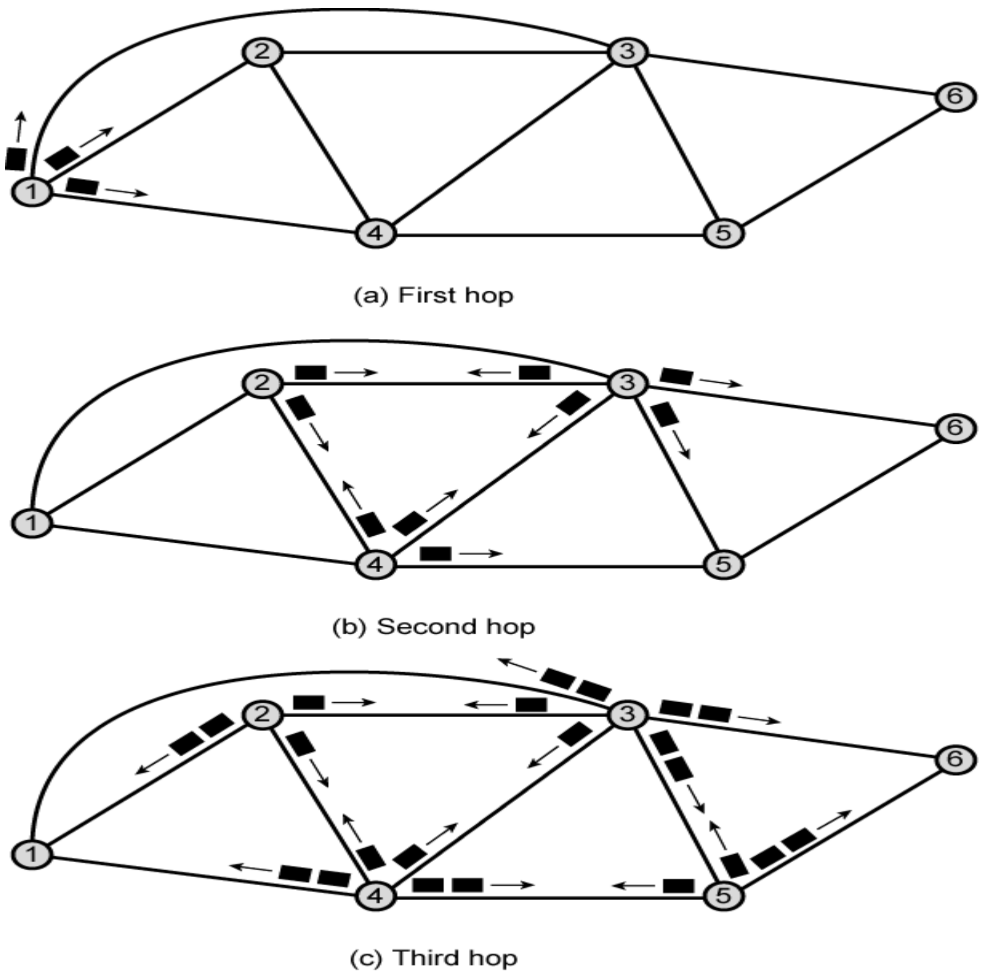


Figure 3.1(a, b, c) Flooding Routing [15]

In link state routing, each node has a complete idea about the topology of the network and the information about the cost of each link. Each node will periodically send the Link State Packet (LSP). This packet basically has information about the cost and the situation of the network routes. The LSP has the following information: ID, age, and sequence number of LSP packets for each route. In the link state, LSPs will be broadcast to all the nodes in the network. After the node receives an LSP packet, its routing table is updated, and this information is useful in building the shortest path between itself and the source node.

In this routing protocol, every node shows neighbor node cost, each node sending an estimated amount of information for the shortest route to the destination for all the connecting neighboring nodes. The receiving node then uses this information to update its own routing table with the shortest path algorithm. This type of routing has strong competition and is easier to run and implement. It doesn't require a large memory for saving routing information [16].

For sources routing, all the packets in the network have complete information about the route needed to reach the destination. The good thing about this type of routing is that, there is no routing loop problem. The main limitation of this routing is the presence of a high delay overhead.

3.3 Classification of the Ad Hoc Routing Protocols

The routing protocols in MANET networks work under three types: proactive, reactive and hybrid.

In proactive routing each node has complete information for the entire connected node in the network. Routing tables are updated every time when the network topology is changed by the mobility of the network nodes. Destination Sequenced Distance Vector (DSDV) [17] is an example of this type.

A reactive routing protocol means that the routes are found for nodes in the network only if, there is a need to send data to a destination. DSR and AODV are examples of reactive routing protocols [17].

Hybrid routing combines the properties of the two previous routing protocols (proactive and reactive) to give more efficiency for routing in MANET. Zone routing protocol is an example of this [17].

3.4 Mobile Ad Hoc Routing Protocols

In this section, AODV and DSR of MANET routing protocols, which were used in the study, are presented.

3.4.1 Ad hoc On demand Distance Vector (AODV)

AODV uses an on demand algorithm, it builds a route between nodes only when it is required by source nodes.

Furthermore, AODV builds trees which connect multicast group members. Trees consist of the group members, and the nodes needed to connect the members. To ensure that freshness routes are used, sequence numbers are used. It is a loop-free, self-starting, and can be scaled to a large number of mobile nodes.

When a source node requires a route to a destination, it broadcasts a route request (RREQ) packet through the network. RREQ packet contains hop count, source and destination address and Destination sequence number (DSN). Nodes receiving this packet update their information for the source node and set up backwards pointers to the source node in the route tables.

If the node has a sequence number greater than or equal to that contained in the RREQ it will send a route reply packet. A node receiving the RREQ may send a route reply (RREP) if it is either the destination, or if it has a route to the destination with a corresponding sequence number greater than or equal to that contained in the RREQ. If this is the case, it is unicasts a RREP back to the source. Otherwise, it rebroadcasts the RREQ if there was link failure. It will rebroadcast the RREQ [18-19].

When the source node receives the RREP, it starts to forward data packets to the destination. If the source receives an RREP with a greater sequence number with the least hop count, it considers this route as the best and begins to forward data packets through it.

If there is a route which has data transmission from the source to the destination, this route is considered active, as long as there are data packets periodically travelling from the source to the destination along that route. If the source stops transmission the data packets, the link will be timeout and be deleted from the source routing table. If a link break occurs while the route is active, the node that detected the link failure will propagates a route error (RERR) message to the source node. If the sources node receiving the RERR, still demands the route, sources will initiate a route discovery again.

Hello messages in AODV are used locally for route maintenance. If two nodes are no longer neighbors then the first node invalidates all routes through second node. The first node may also send an RERR messages to all nodes that used this unveiled route. AODV is using table-driven which means each node has only one hop routing information.

Figure 3.2 shows, the RREQ propagation and RREP path between the sources and the destination in AODV protocol.

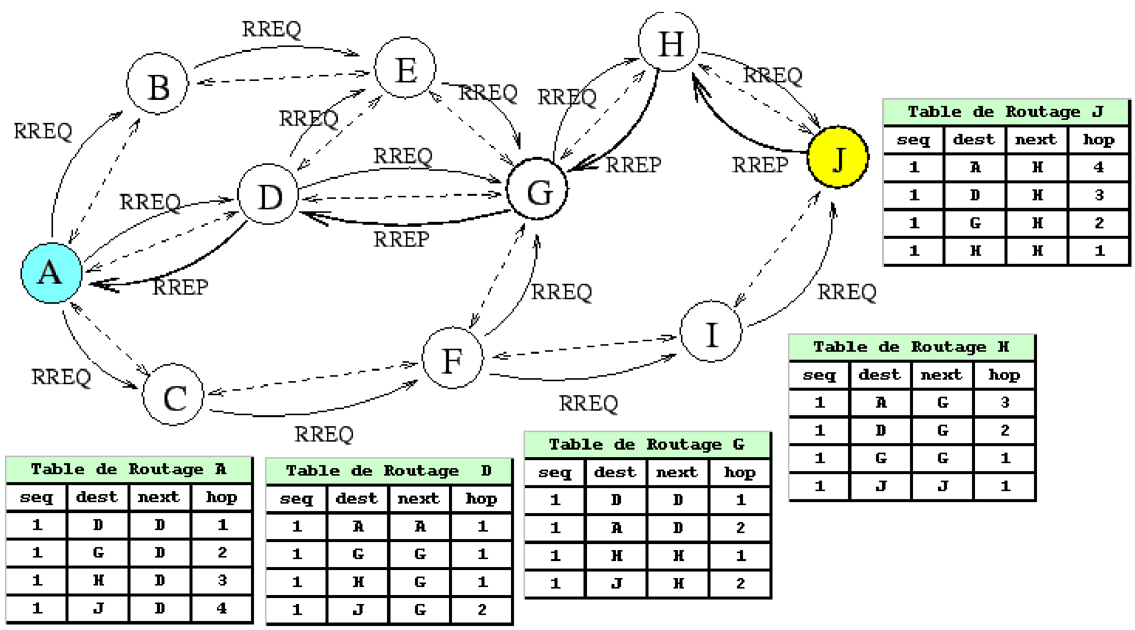


Figure 3.2 RREQ and RERP Path Propagation the Source and the Destination [20]

3.4.2 Dynamic Source Routing protocol (DSR)

This routing protocol is a simple and efficient routing protocol designed specifically for use in the multi-hop wireless ad hoc networks of mobile nodes. Here the network is completely self-organizing and self-configuring, without the need for any existing network infrastructure or administration. DSR was the same as AODV in broadcasting RREQ packets by the source to find the destination with unicasting RREP packet. Each ROUTE REQUEST message identifies the initiator and target of the Route Discovery, and also contains a unique id-request. DSR protocol has two stages of routing "Route Discovery" and "Route Maintenance" which allow for the discovery and the repairing of the routes path between the source and destination through the network. The protocol is on-demand which means that no route occurs until there is a need for the transmission of a data packet through the network. DSR used a route cache or a route record to save the routes in the source node.

DSR doesn’t need to consistently update its routing table like AODV. It also uses a cache memory which decreases the control overhead. DSR works very well in high node density low mobility networks. Figure 3.3 and 3.4 show the route record which was created during a routing discovery time and route replay.

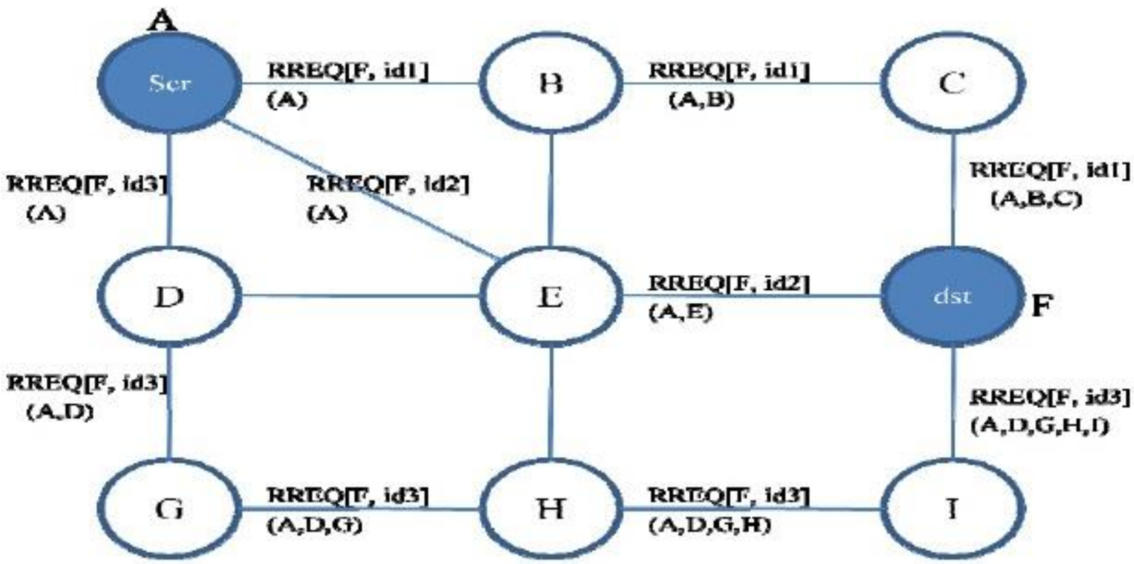


Figure 3.3 Route Record is Created During a Routing Discovery Time [20]

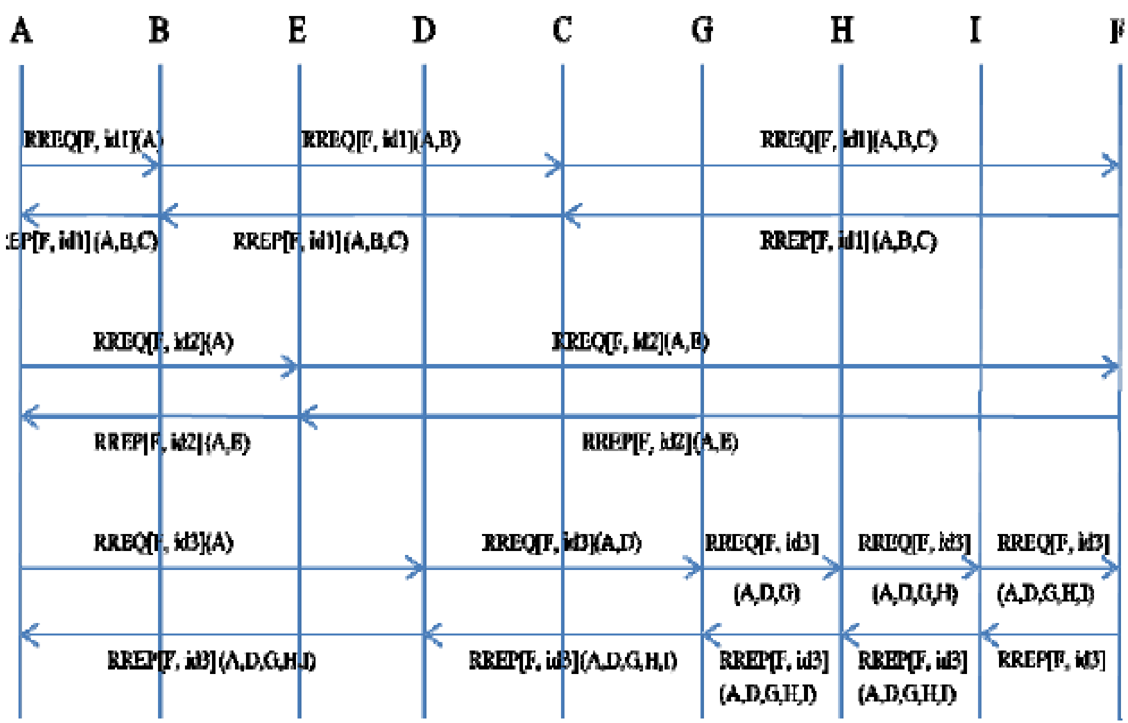


Figure 3.4 Route Replay is Sent from the Destination to the Source Through Route Replay Packet [20]

Chapter 4

VOICE APPLICATION AND CODING SCHEMES

4.1 Overview of VoIP and Protocols

Voice over Internet Protocol (VoIP), is one of the newest technologies in recent years, and uses high-speed broadband internet connections to make voice calling instead of relying on traditional phone lines.

VoIP technology has revolutionized the field of telecommunication and has changed the way we communicate around the world. The enhanced features and the flexibility of the VoIP services have gained the attention of people everywhere, and are available for personal use or for commercial use. VoIP services have proven its significance everywhere.

Through the years, there was an urgent need to improve the performance of communication between mobile devices over different types of topologies in the network. MANET, in particular, requires a good QoS for multimedia applications. Another problem facing VoIP is that of mobility, limited coverage area range, and a high-bandwidth requirement. All these factors can limit the performance of the system since most of its applications are done in real time, and are affected by a loss or delay in the packets.

These limitations have prompted huge research in developing VoIP applications making effective use of available bandwidth. More researchers are developing very good compression algorithms that are flexible in bandwidth utilization and data rate management. By so doing, QoS in coding, decoding, compression, decompression can be guaranteed.

The time taken to convert the voice from the analogue format to the digital format, and voice compression and decompression operations and other voice processing must not consume a lot of processing time.

4.2 VoIP Communication in MANET

Voice-over Internet Protocol (VoIP) is a way that allows you to do voice communication using an Internet instead of an analog phone. VoIP codec protocols doing voice signal conversion and digital compression for analog /digital signal. These codecs differ in their coding and frame rate which affect the speech quality for voice communication through the network.

Before the packet is transmitted over networks, the voice signal has to be digitized at the sender side; the reverse operation happens on the receiver side.

The voice is encoded and then included into packets of equal size. The headers of these packets have the following information: Real-time Transport Protocol (RTP) 12 bytes, User Datagram Protocol (UDP) 8 bytes, Internet Protocol version 4 (IPv4) 20 bytes and the payload. Payload contains the compression voice speech with certain size depending on encoding schemes used [21]. VoIP packet header fields are shown in Figure 4.1.

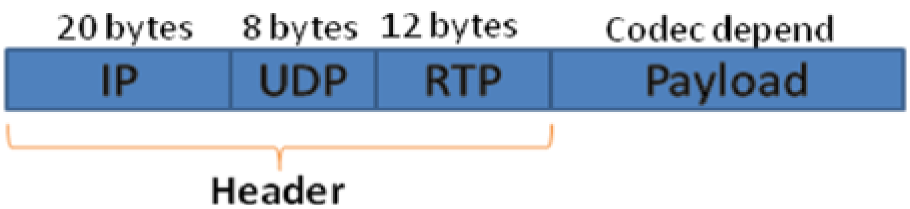


Figure 4. 1 VoIP Packet [21]

Voice packets are transmitted using the route replay path that was established by MANET routing protocol. In MANET, RTP/UDP protocol is used for the transport of the voice packets. Figure 4.2 shows the VoIP system in MANET network.

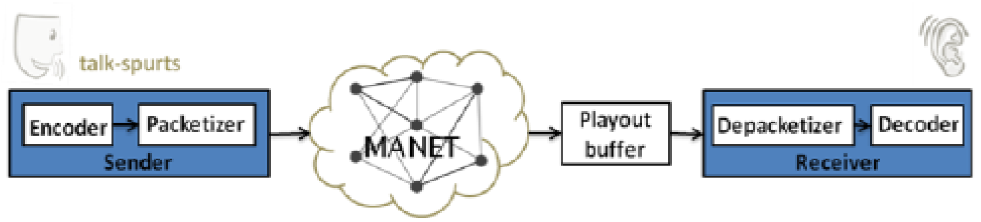


Figure 4.2 VoIP System in MANET Network [21]

4.3 Coding Schemes for Voice Application

Voice applications use several voice coding schemes. The uses of different schemes are dependent on the available bandwidth for the system and the amount of quality required for the service. In this thesis, PCM and GSM voice coding schemes were used.

4.3.1 Pulse Code Modulation Schemes (PCM)

PCM comes from Pulse-Code Modulation**.** PCM is a digital conversation of an analog signal where the magnitude of the signal is sampled at uniform intervals and then quantized. PCM is a coding algorithm which converts the voice signal from analog to the digital shape.

PCM is used by many countries. The most popular PCM coding scheme is G.711, which requires 64 kbps bandwidth and 10 msec frame size. PCM is used in digital telephone systems and is also the standard of the digital audio in computers. On the other hand, a high quality of voice transmission is required as well as a high bandwidth. As a result, most countries have stopped using it and prefer other voice coding schemes [22]. Figure, 4.3 shows Pulse Code Modulation (PCM) schemes coding processes

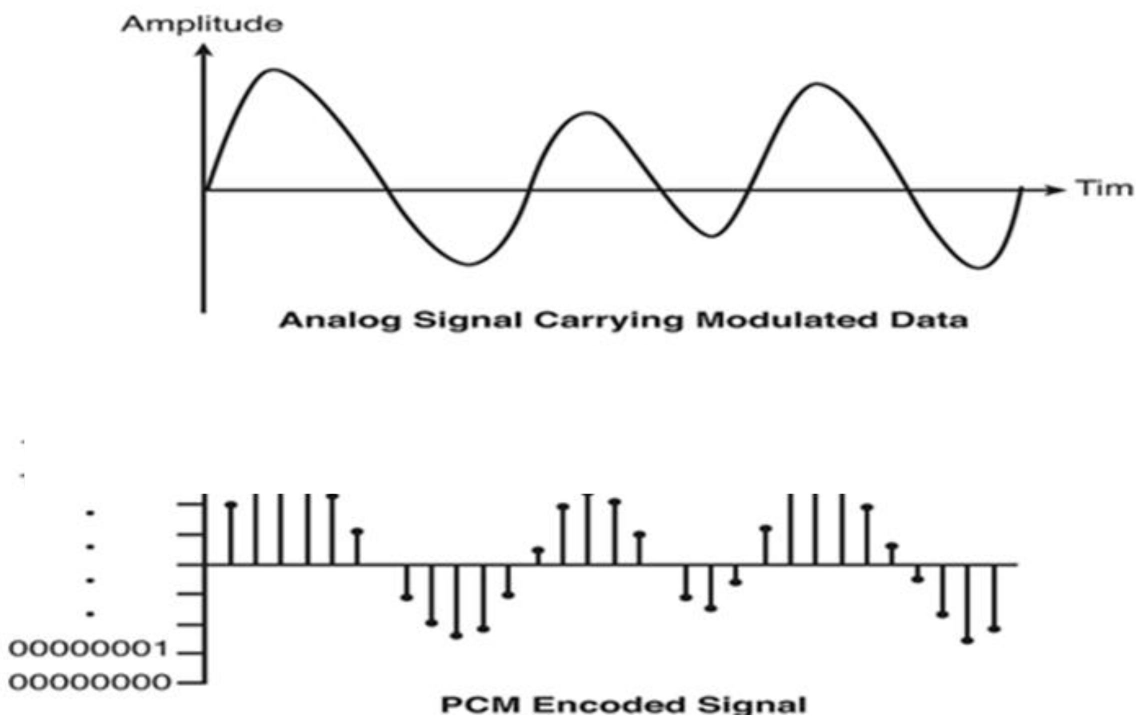


Figure 4.3 Pulse Code Modulation (PCM) Schemes Coding Processes [22]

4.3.2 Global System Mobile (GSM)

GSM means Global System for Mobile Communications, and is a standard developed by the European Telecommunications Standards Institute (ETSI) to describe protocols for second generation (2G) digital cellular networks used by mobile phones.

This standard was developed to replace the first generation (1G) analog cellular networks. it was developed to content data communications, through a circuit switched network. Well known examples are General Packet Radio Services (GPRS) and Enhanced Data rates for GSM Evolution (EDGE) or EGPRS technologies [23].

GSM has a low bandwidth with only 12Kbit/sec with 20 msec frames size with reasonable voice quality.

4.4 Quality of Services for Voice Application

4.4.1 Bandwidth Requirement

The bandwidth is always a limiting factor. However, a good amount of bandwidth will help to propagate the voice packet to the destination without any collision or delay. The amount of bandwidth required for voice applications is different and also depends on the voice schemes being used. One of the challenges faced with voice transmission is to design a voice transmission network making the best use of the available bandwidth.

4.4.2 Compression Method

Another element for a good voice application is the compression algorithm. A good compression method which retains the quality of the voice after the application of the compression algorithm and doesn’t consume too much CPU's time during the compression /decompression process of voice packets should be used. An example of the voice compression method is Adaptive Differential Pulse Code Modulation (ADPCM) and Adaptive Transform Encoding (ATC) [24-25]

4.4.3 Jitter

In VoIP, jitter is the difference in the arrival time of the packet caused the congestion of routing overhead. Jitter buffer can be used between the transmitter and the receiver to handle jitter. To accomplish this, a time stamp is used to play the packet at the original or correct time point. Furthermore, sequence numbers will be utilized to reorder the disordered packets at the receiver to get the an original sent packet. A resend request will be made for any detected lost packets [25].

Chapter 5

SIMULATION SETUP

5.1 OPNET Simulation Environment

OPNET is a commercial software that provides performance analysis of computer networks and applications. It stands for Optimized Network Engineering Tools. OPNET is one of the best and most powerful networks modeling software. It provides an excellent tool for networking and is very easy to use. OPNET has a vast content library, with all the elements that a designer needs for building any type of network no matter how complicated it may be. All elements that are required to build and measure the simulation are available in OPNET.

Most of the tools that a designer needs are in the OPNET library. For example, the time to begin work for the application in the server or the profile with the client can be easily specified using OPNET. In addition, it provides mobility elements: speed, area of movement, start time movement, the pause time of the mobile node and their corresponding probability type (e.g. uniform, exponential, etc.). The number of the performance metrics provided in OPNET library is enormous. It is classified in a very convenient and intelligent way for all network types. OPNET provides an easy way to simulate all types of networks, and it allows the user to discover any problems that might occur speedily and efficiently. All the applications and time setting are available with their corresponding distributions. The negative side of this program is that it requires more computer time because of the huge numbers of runs required to collect simulation results. It is also expensive can cost anything between 6000 USD and 25000 USD.

The following steps are required to build a simulation in OPNET.

1. **Create network modeler**

The first step is to create the required network. This can be done by building your network environment. Enter all the appropriate parameters that specify your network type and topology.

2. Choice performance metrics

The second step is to choose the performance metrics that are required for the evaluation of the simulation work.

3. **Run your simulation**

Having completed the first two steps the user is now able to run the simulation of the network

4. **Study the results and analyze it**

This is the final step in the simulation program. Study all the resulted graphs and data and use the necessary methods to explain and analyze them. The resulting graphs and data can then be studied and analyzed. Figure 5.1 shows all these steps

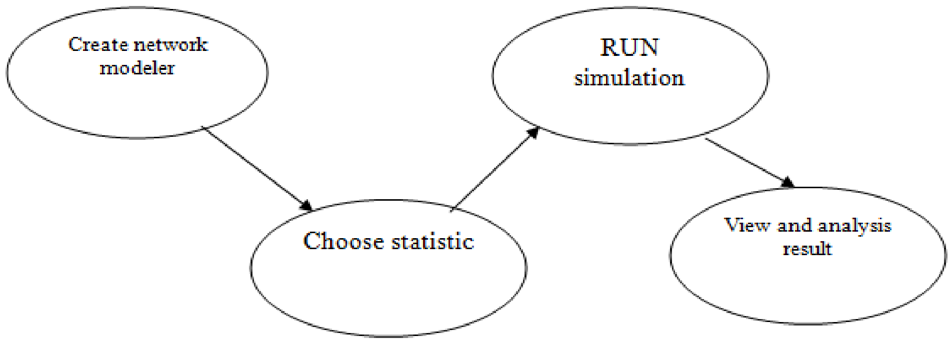


Figure 5.1 Basic Steps to Design and Build Simulation using OPNET Program.

5.2 Algorithm Used for Simulation

In this section, the algorithm to build and design the simulation is explained. A simulation was built for a mobile wireless ad hoc network with two types of MANET routing protocols AODV and DSR. Two protocols were compared to determine which one was more suitable based on the results of the simulation. In addition, two voice application schemes, PCM and GSM, were utilized as provided by the OPNET library. Finally, the results of the simulation which was conducted using a different number of clients were analyzed. Figure 5.2 shows algorithm for simulation.

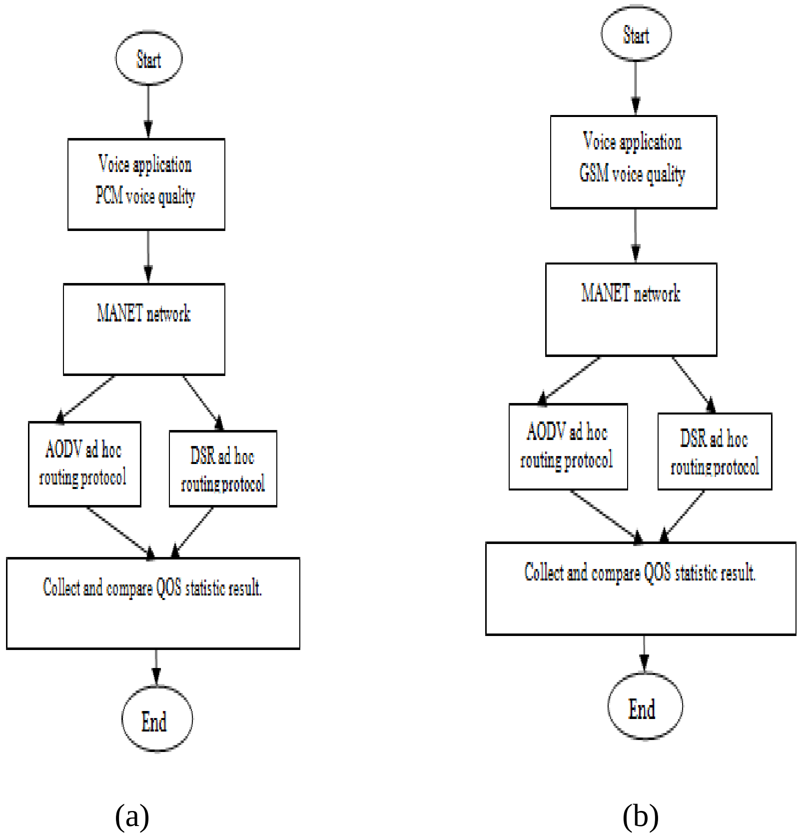


Figure 5.2 (a, b) Two Algorithms used in Simulation

5.3 Simulation Setups in OPNET

In this simulation, OPNET version 17.1 was used to design and build wireless ad hoc networks. The first setup is to specify all the network elements (name of the project, the type of network and its coverage area). In this simulation, the network type is assigned as a campus network with a coverage area of (5x5) kilometers [14] as shown in Figures 5.3 and 5.4.

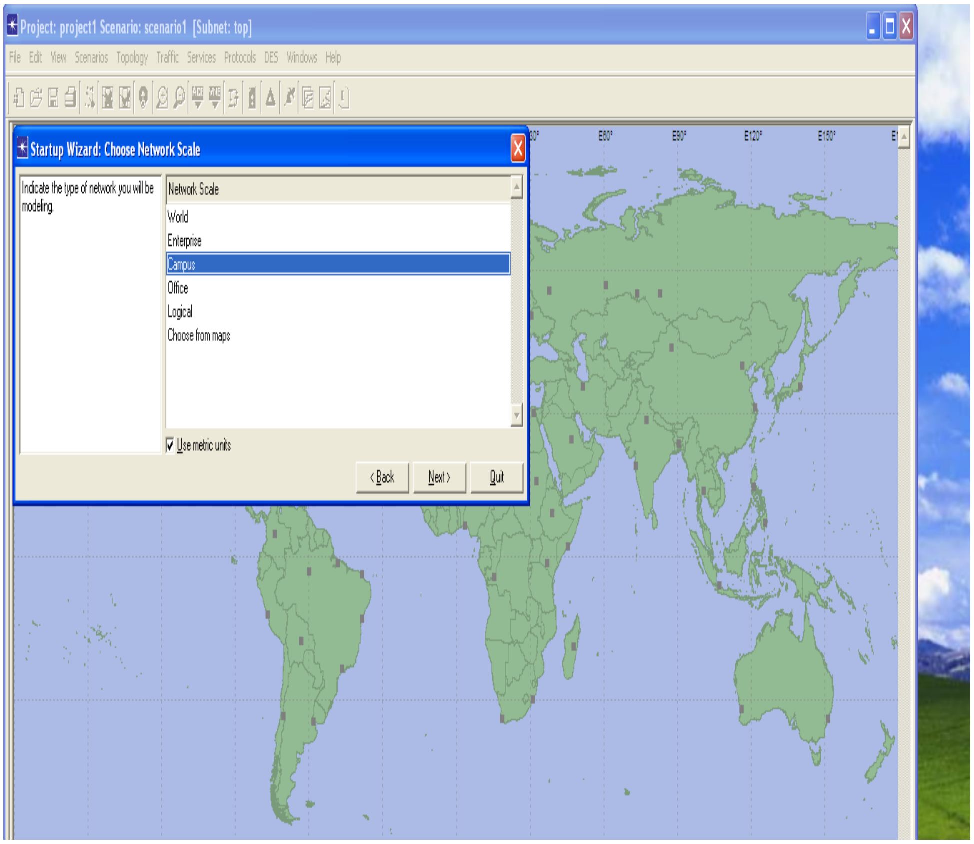


Figure 5.3 The Assign of the Network Type

Next, the components required for the design of the ad hoc environment were configured. After that, a number of components were defined for example; application definitions, profile definitions and mobility profiles. The scenarios were built with 25, 50, 75, and 100 nodes and 1, 12 clients was used in each scenario which were distributed across the network and communicated only with one server.

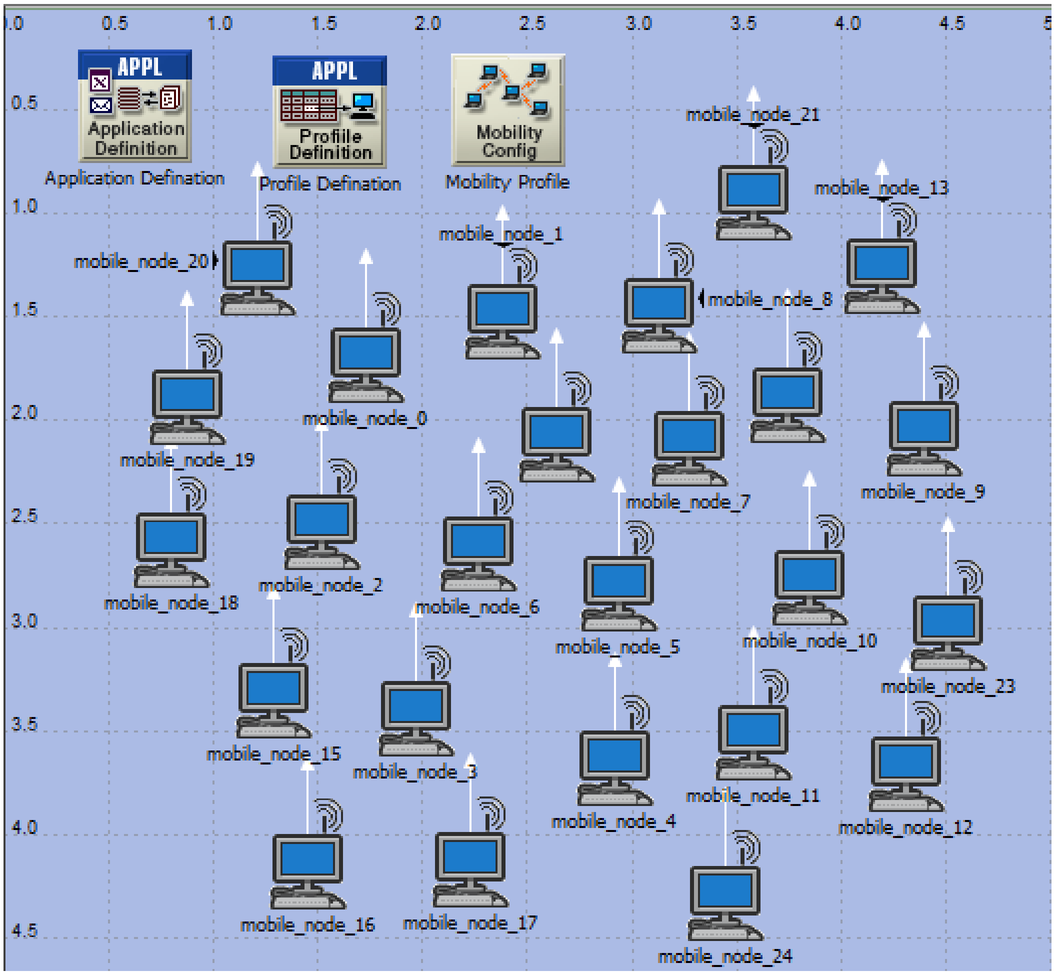


Figure 5.4 Ad Hoc Network with 25 Nodes

Table 5.1: Parameters Value for the Network

Network parameter	**Value**
Network size	5 km x 5 km
Network type	Campus
Network family	MANET
Number of nodes	25, 50, 75, 100
Element of the network	Application Definition, Application Profile definition, Mobility profile, workstation node (client and server)
Simulation time and seed values	1000 sec with seed values (128, 130, 132, 134)
Number of the client nodes	1, 12
Number of the server	1

5.4 Voice Communication in OPNET

A voice application enables two callers (called node with PCM/GSM application definition, and other calling nodes with profile definition of PCM/GSM voice application) to establish a virtual channel over which they can communicate using digitally encoded voice signals.

UDP is transport protocol used for the voice application. The voice packets are sent over Real-Time Protocol (RTP).

Figure 5.5 shows the voice communication between two mobile nodes.

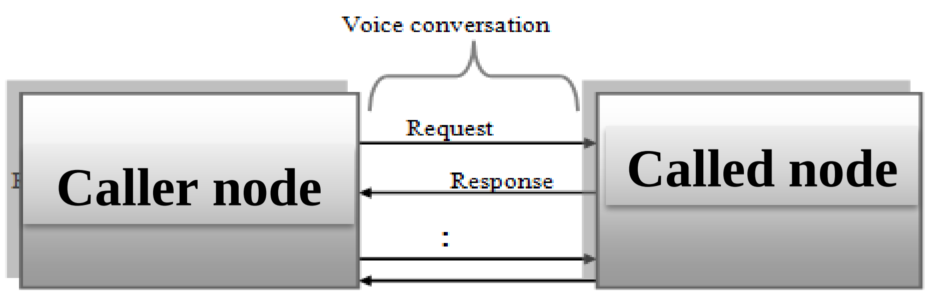
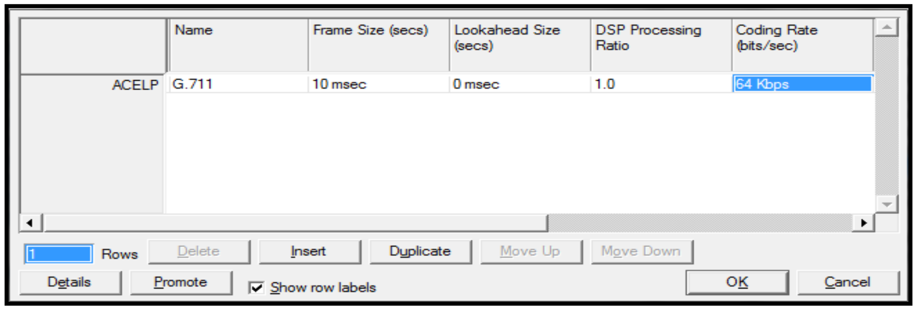


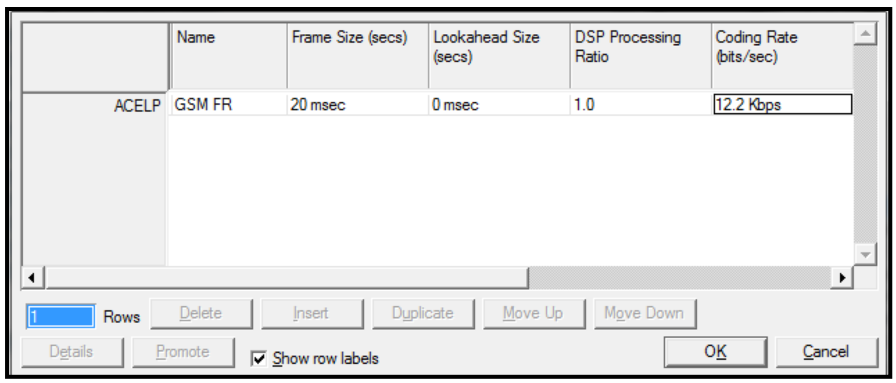
Figure 5.5 Voice Communication with One Client (caller) and One Server (called) node

5.5 Application Settings for Voice

A voice application was assigned for the designed network. The voice application included two schemes PCM with voice encoding for G.117 and GSM corresponding to the voice encoding for GSM-ER as shown in Figure 5.6 (a, b). These two voice application schemes were assigned for each MANET routing protocol AODV and DSR. Calculations and analysis were done for the performance metrics of these two voice application schemes over AODV and DSR.



(a)



(b)

Figure 5.6 (a, b) Voice Schemes Setting for PCM and GSM Respectively

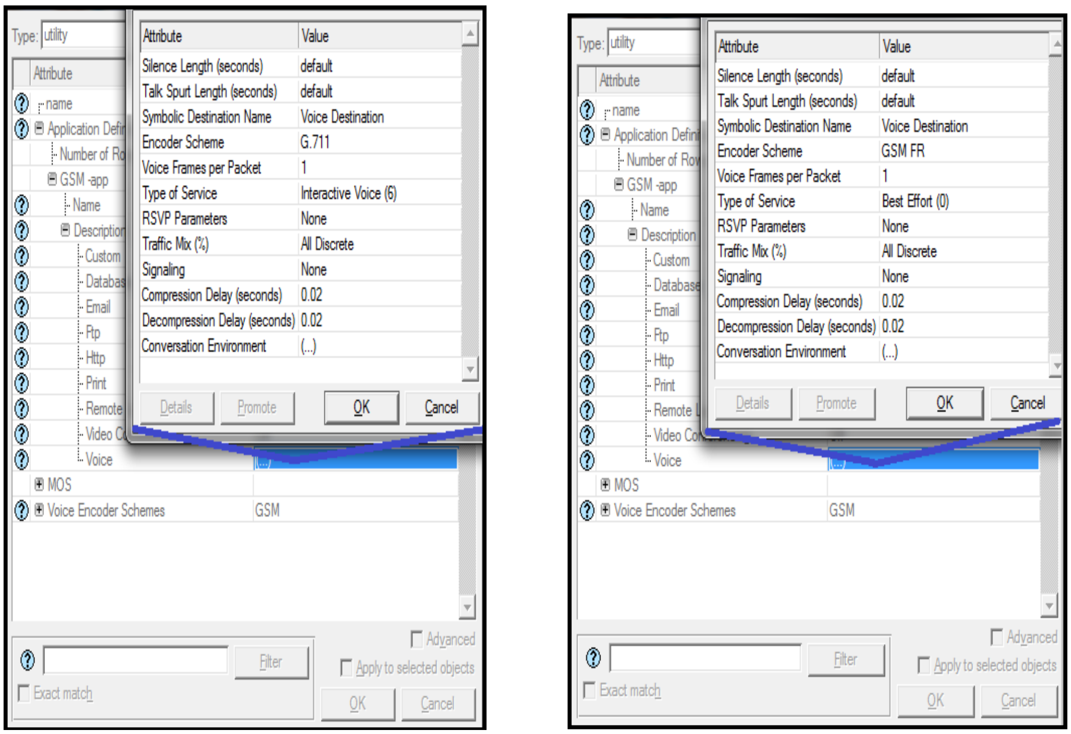
Table 5.2: Parameters of PCM Encoding Scheme

Parameter of PCM voice application	**Value**
Application	Voice
Voice encoding schemes	PCM (G.711)
Frame size	10 msec
Coding ratio (kbits/sec)	64 kbit/sec

Table 5.3: Parameters of GSM Encoding Scheme

Parameter of GSM voice application	**Value**
Application	Voice
Voice encoding schemes	GSM (GSM-FR)
Frame size	20 msec
Coding ratio (kbits/sec)	12 kbit/sec

G.711 requires 64 kbps bandwidth and a10 msec frame size. GSM has low bandwidth only 12Kbit/sec with 20 msec frames size and reasonable voice quality. PCM has a larger bandwidth compared to GSM [26].



(a) (b)

Figure 5.7 (a, b) Some Important Voice Application Attributes for PCM and GSM

Here is the explanation of same important voice application settings that is shown in Figure 5.7

- Encoder Scheme: Encoder Scheme was used by the calling and called sides. For (PCM/GSM)

- Number of voice frames per packet: defines the number of frames included in each voice packet.
- Compression and decompression delay (seconds): this specifies the delay in compressing and decompression of a voice packet [26].

5.6 Profile Settings for Voice

A profile is used for each client node that is supports the voice application. The server which supports the voice application cannot communicate with the client unless the client has already been identified and can support the application on the server. The profile works as an intermediate between the server that supports the application and the client. Furthermore, a profile is used to determine all the timing parameters that will decide the provision of the services, the number and duration of a request for a service and the pattern of the communication (synchronize, series). Figure 5.8 shows the parameters used in the simulation for PCM profile settings. The Same settings are also valid for GSM profile with a different name [27].

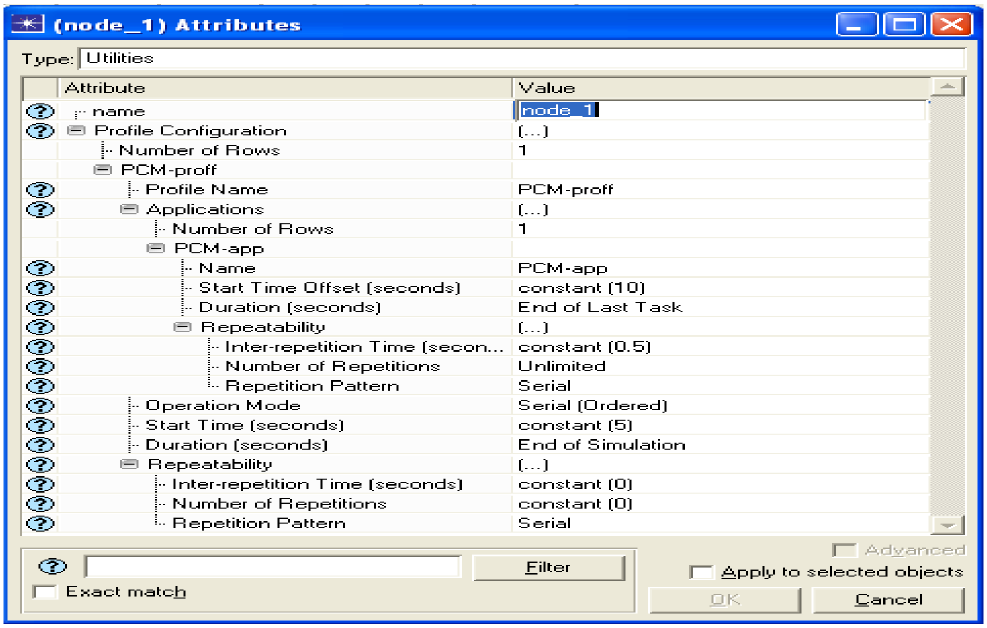


Figure 5.8 Voice profile setting (PCM)

Table 5.4:Parameters of Voice Application Profile for PCM and GSM

Network parameter	**Value**
Profile name	PCM-prof, GSM-prof
Application name	PCM-app,GSM-app
Start time offset of application (Sec)	Constant (10)
Duration of application	End of the last task
Number of repeating of application	Unlimited
Start time of profile (Sec)	Constant (5)
Duration of profile	End of the Simulation
Number of repeating of application	Once at time

Figure 5.8 and Table 5.4 show the profile of PCM voice application scheme. Same settings are valid for GSM as well. The application name is (PCM-app). The application starts to run after 10 seconds from the beginning of the simulation time until it has completed its last task. The next request is sent after 0.5 seconds. This is repeated periodically during the simulation time. The application and profile are run in serial mode. The profile is run after 5 seconds and there is no profile repetition. The profile is a continuously running until the end of the simulation time.

5.7 Mobility Settings for Mobile Nodes

In the simulation, a random waypoint mobility model was used for mobile nodes as shown in Figure 5.9. In this algorithm, the node will first choose a random position to move to it. This random position must be in the movement area within the algorithm boundary. Then nodes will move into that position at a certain speed. The mobile node will wait for a definite amount of time [24-25]. Then it will select a new random position and move to it. Figure 5.9 shows the random waypoint algorithm for 1000m*1000 m mobility area.

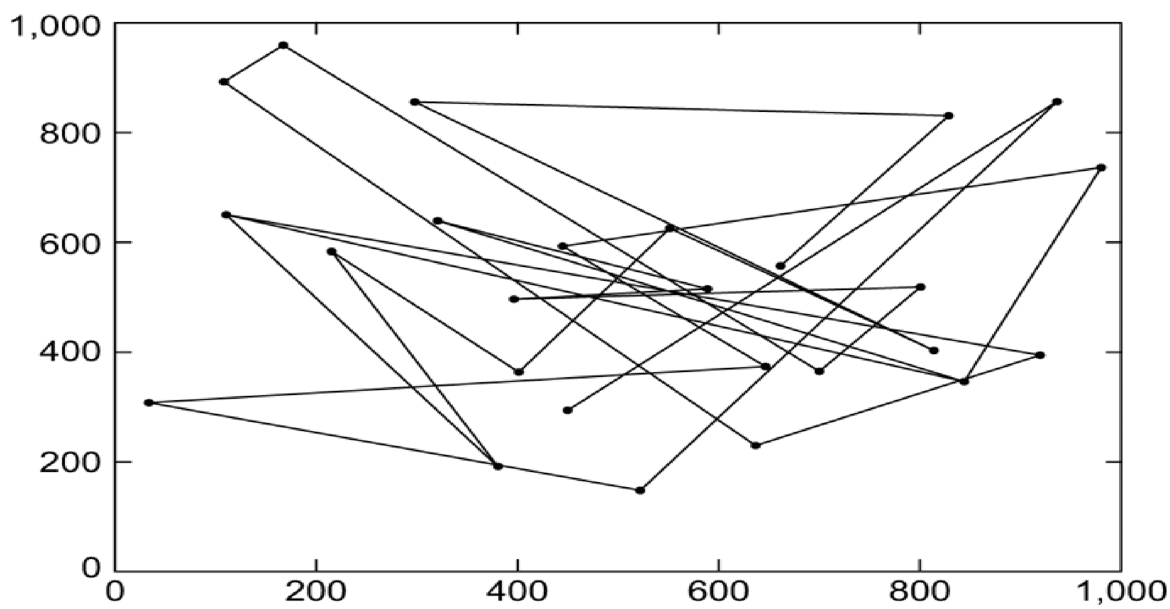


Figure 5.9 Random way Point Mobility Algorithm [28 -29]

The parameters (x-min, y-min, x-max and y-max) are used to define the boundary in meters (500x500). Speed is defined as the speed of the mobile node and set with 1 meter per second. The pause time here is set to 5 seconds. So nodes will wait this much time before moving to the next random station. The mobility nodes start time is set to 10 sec after the beginning of the simulation time. The mobility of nodes will run continuously until the end of the simulation time. Figure 5.10 and Table 5.5 show, the mobility profile settings.

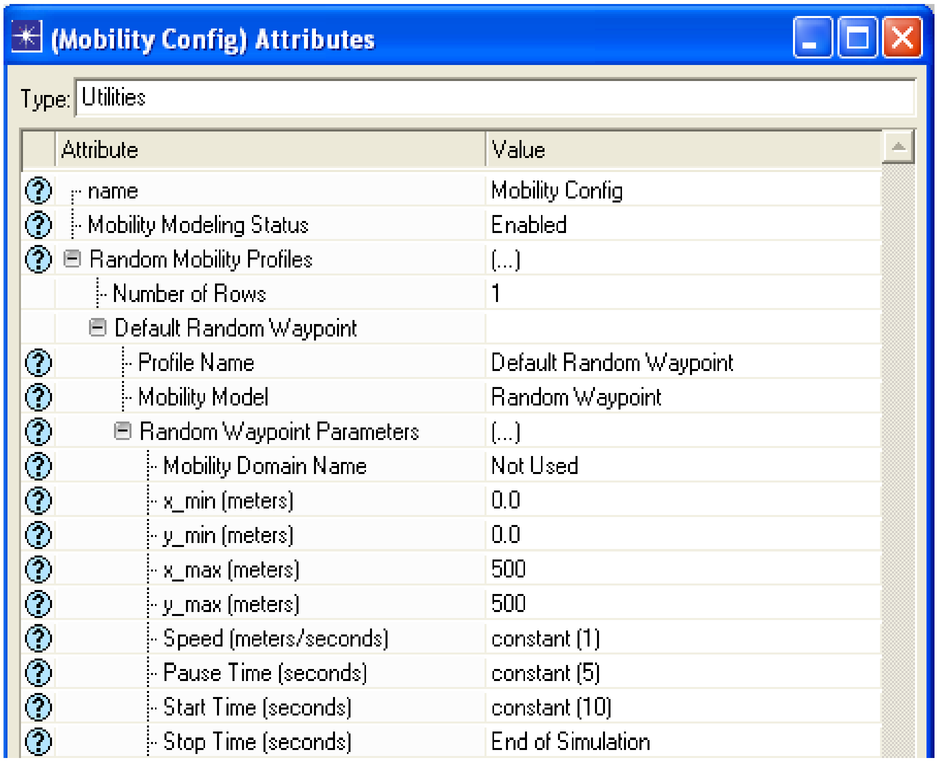


Figure 5.10 Mobility Setting for The Mobile Node

Table 5.5: Mobility Parameters of Mobile Nodes

Network parameters	**Value**
Speed (m/s)	Constant (1)
Pause time (Sec)	Constant (5)
Start time of movement (Sec)	Constant (10)
Stop time of movement	End of simulation time
Movement area	500 m*500 m

5.8 Mobile Workstation Settings

A mobile workstation (client or server) moves with random waypoint mobility model with parameters shown in Figure 5.10.

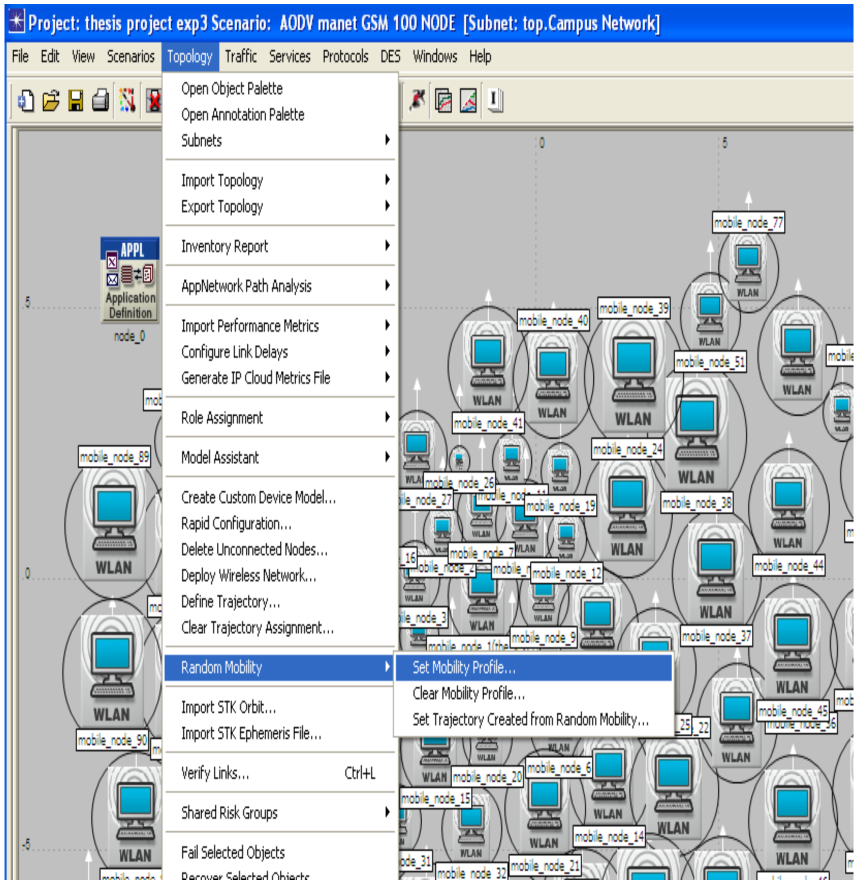


Figure 5.11 Assignment of Mobile Profile for Nodes in the Network

Figure 5.11 shows the settings of the mobility profile. All mobile nodes will have an automatic mobile profile. This mobile profile contains all the information regarding the mode of the mobility (random waypoint and direction movement…etc.) and the speed of the node, and all the related parameters for mobility movement. All the mobile nodes are communication using IPV4 auto address within the network.

Figure 5.12 shows the use of an application deployment wizard. An application for the client and the server is defined as shown in Figures 5.12 and 5.13 for GSM voice application and profile(i.e. mobile node -1 was defined as server and mobile node -10 was defined as client). Same settings were used for PCM voice encoding schemes as well [26].

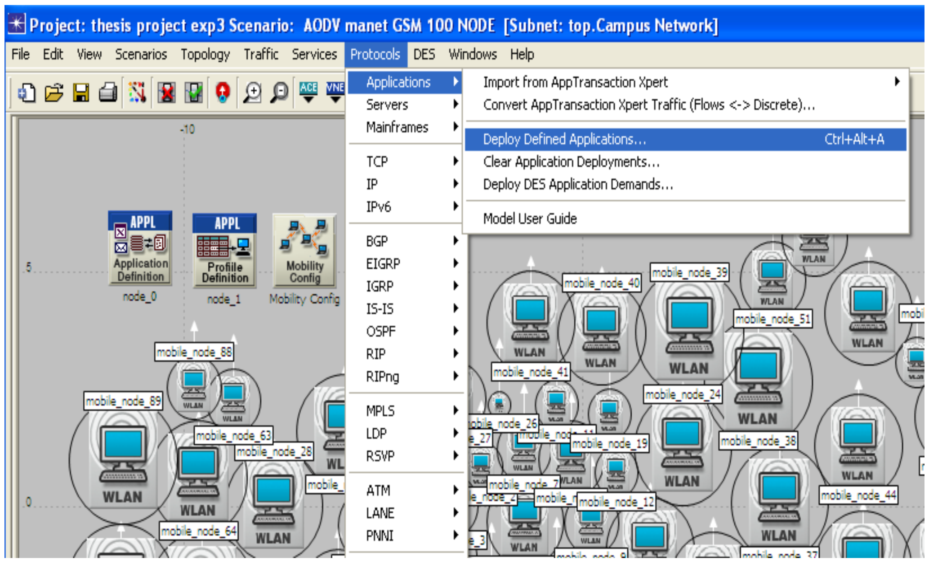


Figure 5.12 Deployment of Application Wizard for Nodes in the Network

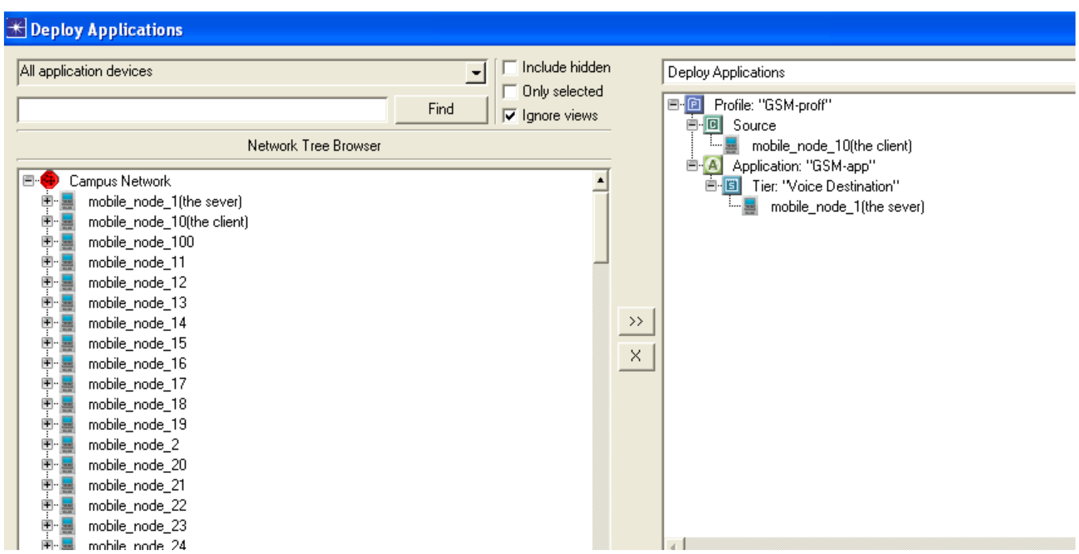
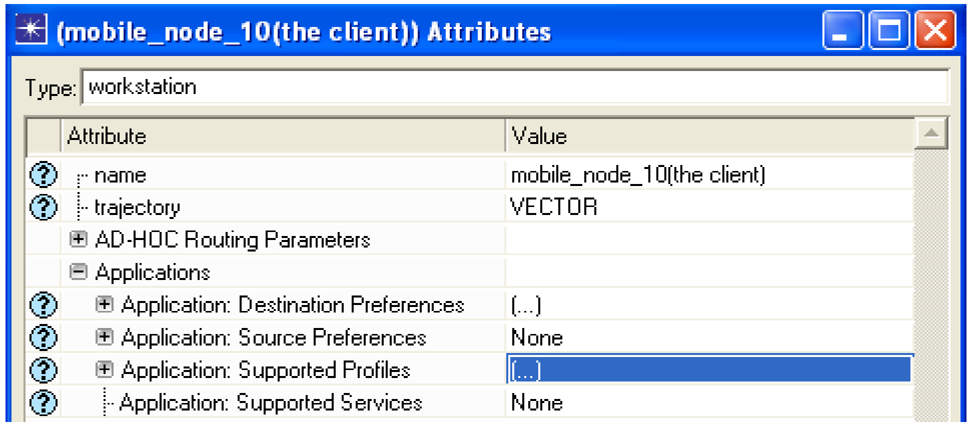


Figure 5.13 Application Deployment Wizards

For a client, two steps are needed. The first step is to set the GSM voice application profile for the corresponding client. This can be done automatically using an application deployment wizard as shown in Figure 5.14 (a, b).



(a)

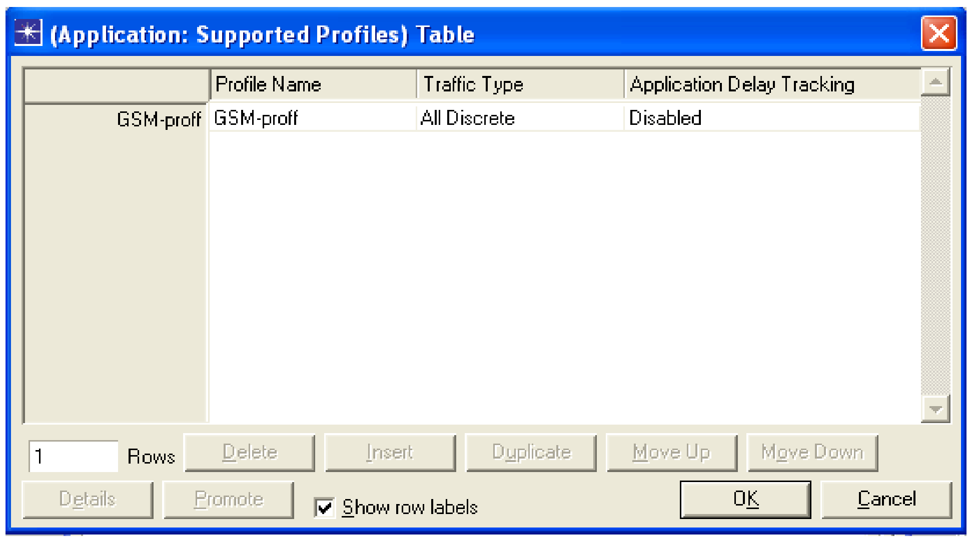


Figure 5.14 (a, b) GSM Voice Application Profile Settings for a Client

The second step is to establish the communication between the server node that already supports the application and the client which supports the profile of application. This can be done using the application destination preferences. In this case, it is called voice destination. It provides more than one client contact with one server using a virtual name as shown in Figure 5.15 (a, b).

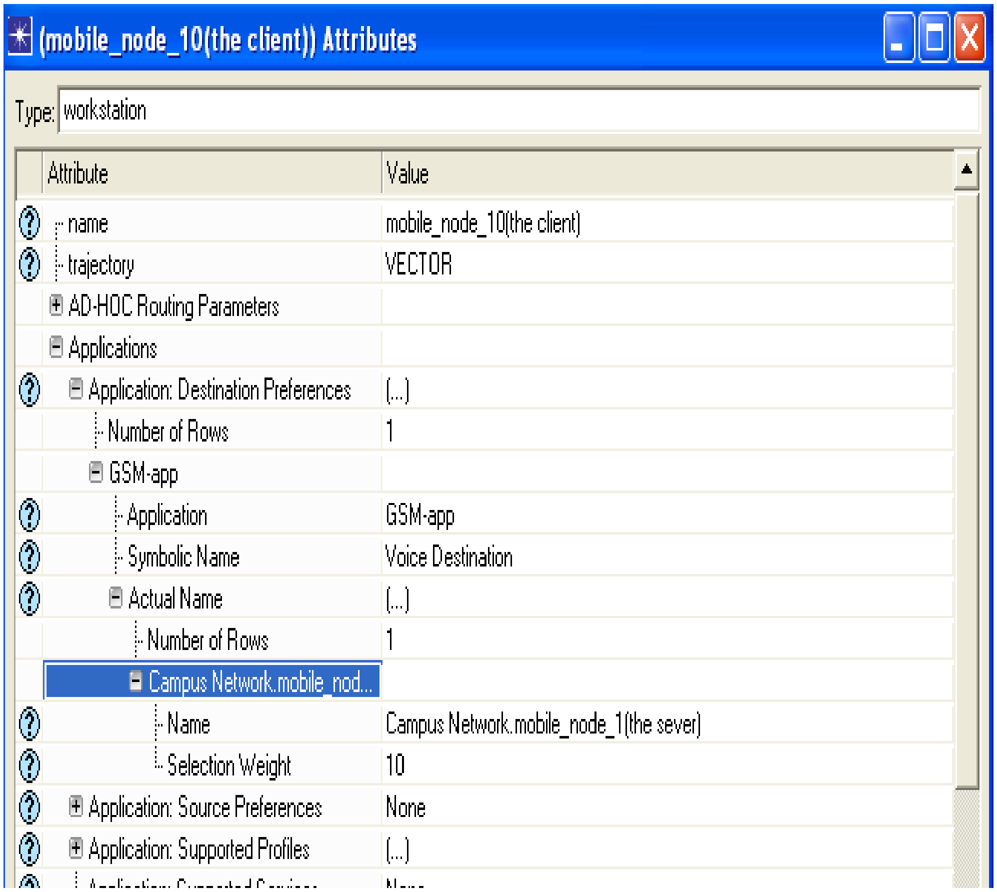


Figure 5.15 Communication Setting Between the Client and Server

The next step is assigning MANET routing protocol, which is used to define the routing movements as shown in Figure 5.16.

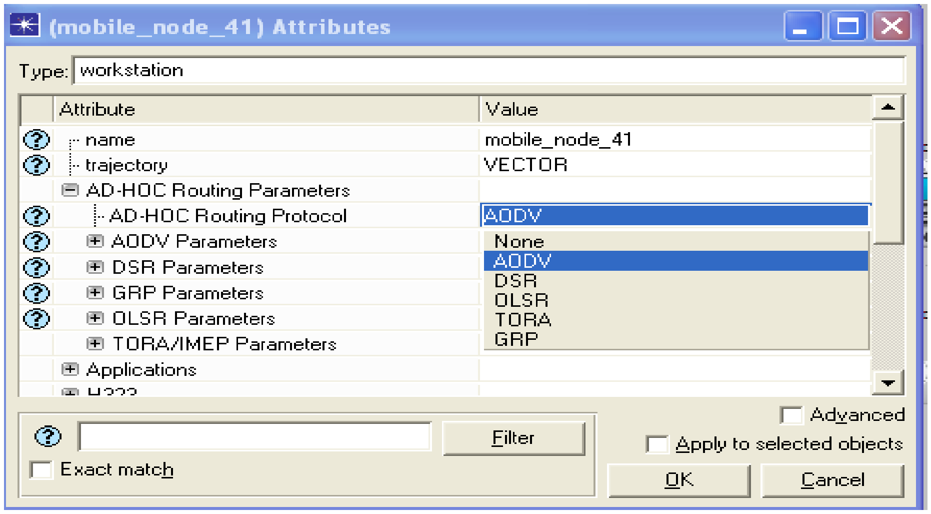


Figure 5.16 Assignment of MANET Routing Protocol

From the ad hoc routing protocol list, we can choose one of the MANET routing protocols. A physical layer character direct sequence IEEE 802.11b with date rate 11Mbps is selected which transmits power 0.005 W as shown in Figure 5.17

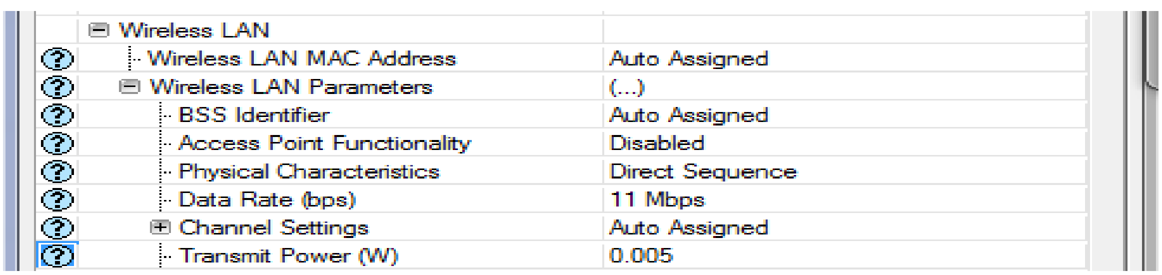
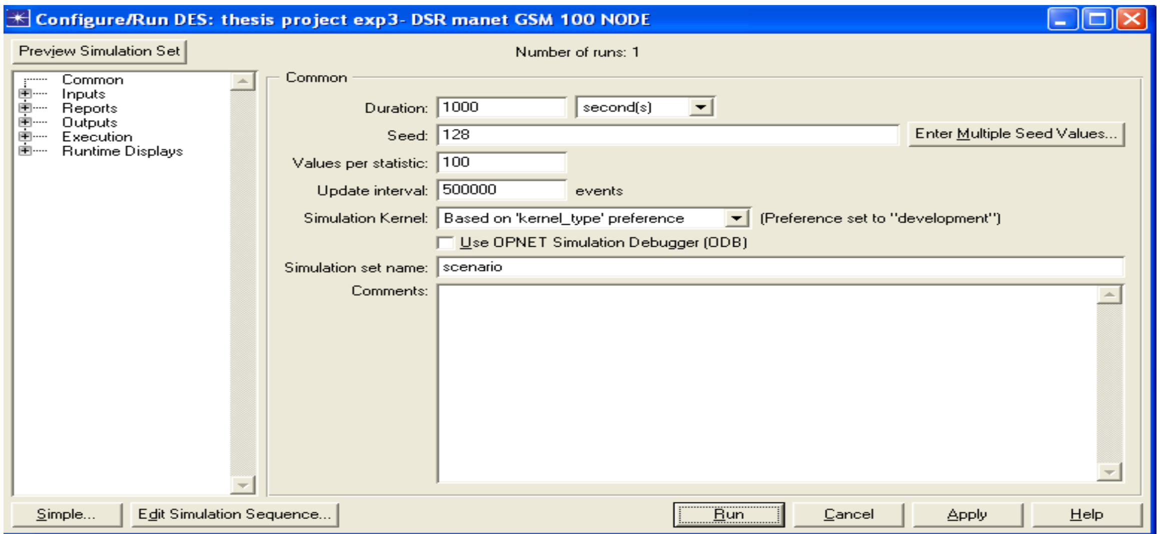


Figure 5.17 Assignments of WLAN Parameters

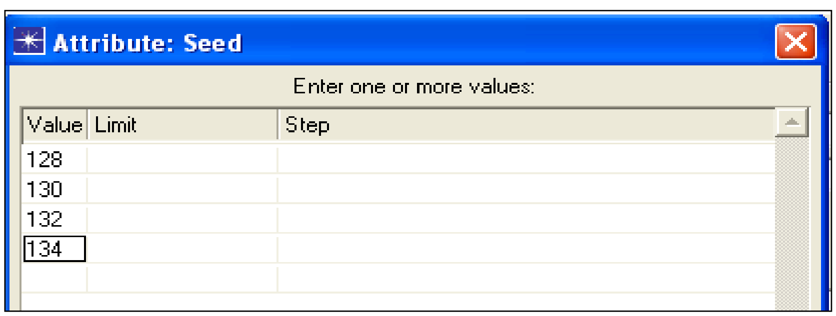
5.9 Choose of the Performance Metrics and Run of the Simulation

The final step is to collect the simulation results for the performance metrics. Choose the individual statistics and assign the voice performance metrics that is required for evaluating performance of MANET routing protocols and voice application.

Different values of seeds are used to generate different random values for four different runs. After that, the average is taken for each performance value with simulation time 1000 sec as shown in Figure 5.18 (a, b).



(a)



(b)

Figure 5.1 (a, b) Different Seed Values and Simulation Run Configuration

5.10 Explanation of the Performance Metrics

In this simulation, two classes of performance metrics were used to evaluate the performance of the ad hoc network. The first performance metric is the route discovery time for the routing protocol. The jitter, packet end-to-end delay, traffic received and traffic sent performance metrics were used for the voice application.

From the last two, the traffic delivery ratio can be created.

Here is the definition of these performance metrics.

- **Route discovery time**

This performance metric represents the time required to discover destination among the intermediate nodes [31].

- **Jitter (msec)**

This represents the delay for voice packets. For example, if two packets moves from the sources with time (t1 and t2), and packets are received with time (t3 and t4) at the destination .The Jitter can be identified as

$$\text{Jitter} = (t4 - t3) - (t2\text{-}t1)$$

- **Packet end-to-end delay (sec)**

It is the total delay time starting from the analog form of the voice at the source until conversion of the analog form at the destination. It includes network delay, encoding and decoding delay, compression and decompression delays [32].

- **Traffic delivery ratio**

It was calculated using equation below:

$$\textit{Traffic delivery ratio} = \textit{Traffic received} / \textit{Traffic send} \times 100$$

Traffic received is the average number of packets received by all the voice applications from the network. Traffic sent is the average number of packets sent from all voice applications to the network [31-32].

Chapter 6

SIMULATION RESULTS AND DISCUSSIONS

6.1 Simulation Results

The evaluation of voice application using two different routing protocols AODV and DSR were done with different scenarios. We have two voice schemes PCM and GSM, over different numbers of the clients (1, 12). And we have used a different number of the nodes (25, 50, 75 and 100). Table 5.6 shows the summary of simulation parameters. Table 5.6 shows a summary of simulation parameters.

Table 6.1: Simulation Parameters

AODV								**DSR**							
PCM with 1and 12 clients				**GSM with 1and 12 clients**				**PCM with 1and 12 clients**				**GSM with 1and 12 clients**			
25	50	75	100	25	50	75	100	25	50	75	100	25	50	75	100

Tables below show the simulation results for route discovery time, jitter, packet end-to-end delay, traffic sent and traffic receive. Performance metrics with given parameters PCM and GSM encoding schemes were evaluated for both AODV and DSR routing protocols with different number of nodes and participating clients in the network.

Table 6.2: Simulation Results for AODV Routing Protocol with PCM voice Scheme for One Client

Performance metric	**Number of nodes**			
	25	50	75	100
Route discovery time (sec)	1.921	1.851	1.701	1.22
Jitter (msec)	0. 1	0.5	1.0	0. 5
Packet end- to- end delay (sec)	0.704	0.572	0.539	0.417
Traffic received (packet/sec)	59.721	82.76	77.325	72.053
Traffic sent (packet/sec)	133.390	144.707	141.991	139.622

Table 6.3: Simulation Results for AODV Routing Protocol with PCM voice Scheme for 12 Clients

Performance metric	**Number of nodes**			
	25	50	75	100
Route discovery time (sec)	3.593	0.670	0.875	0.990
Jitter (msec)	2	1	2.0	1.5
Packet end -to- end delay (sec)	1.859	2.336	1.352	1.452
Traffic received (packet/sec)	57.764	81.769	93.800	89.116
Traffic sent (packet/sec)	1284.981	1271.311	1287.52	1293.22

Table 6.4: Simulation Results for AODV Routing Protocol with GSM Voice Scheme for One Client

Performance metric	**Number of nodes**			
	25	50	75	100
Route discovery time (sec)	2.305	1.299	2.074	1.794
Jitter (msec)	5.5	0.3	0.5	1.5
Packet end -to- end delay (sec)	1.471	0.645	1.106	1.196
Traffic received (packet/sec)	141.852	202.121	186.471	175.657
Traffic sent (packet/sec)	339.502	360.274	356.169	354.488

Table 6.5: Simulation Results for AODV Routing Protocol with GSM Voice Scheme for 12 Clients

Performance metric	**Number of nodes**			
	25	**50**	**75**	**100**
Route discovery time (sec)	4.059	1.861	1.250	1.576
Jitter (msec)	7	2.6	2.1	3
Packet end -to- end delay (sec)	1.765	0.963	2.130	1.961
Traffic received (packet/sec)	102.052	196.959	206.153	187.771
Traffic sent(packet/sec)	3194.729	3198.547	3210.433	3205.397

Table 6.7: Simulation Results for DSR Routing Protocol with PCM Voice Scheme and for One Client

Performance Metric	**Number of nodes**			
	25	**50**	**75**	**100**
Route discovery time (sec)	9.959	9.997	9.685	8.893
Jitter (msec)	4	1	1	0. 8
Packet end -to- end delay (sec)	3.610	2.688	2.484	1.124
Traffic received (packet/sec)	36.83	38.0345	73.27925	70.568
Traffic sent(packet/sec)	132.817	123.055	138.221	136.644

Table 6.8: Simulation Results Data for DSR Routing Protocol with PCM Voice Scheme for 12 Clients

Performance Matrices	**Number of nodes**			
	25	**50**	**75**	**100**
Route discovery time (sec)	12.200	13.237	10.827	10.492
Jitter (msec)	11	3.3	4	2.5
Packet end -to- end delay (sec)	4.921	4.329	2.945	1.997
Traffic received (packet/sec)	25.666	22.537	61.838	89.524
Traffic sent(packet/sec)	1277.164	1239.048	1305.105	1313.988

Table 6.9: Simulation Results for DSR Routing Protocol GSM Voice Scheme for One Client

Performance metric	**Number of node**			
	25	**50**	**75**	**100**
Route discovery time (Sec)	10.781	11.107	8.971	8.849
Jitter (msec)	11	2	2	0.8
Packet end to end delay (Sec)	5.339	3.065	3.821	1.654
Traffic received (packet/Sec)	112.512	71.141	123.835	167.403
Traffic sent(packet/sec)	337.067	298.376	336.709	342.421

Table 6.10: Simulation Results for DSR Routing Protocol GSM Voice Scheme for 12 Clients.

Performance metric	**Number of node**			
	25	**50**	**75**	**100**
Route discovery time (sec)	12.337	12.972	10.085	8.616
Jitter (msec)	23	3	2.5	4
Packet end- to-end delay (sec)	7.418	5.634	4.120	2.561
Traffic received (packet/sec)	37.098	39.683	109.744	157.879
Traffic sent(packet/sec)	3171.163	3096.369	3259.448	3287.749

Figures 6.1 - 6.4 show graphs of the route discovery time performance metric.

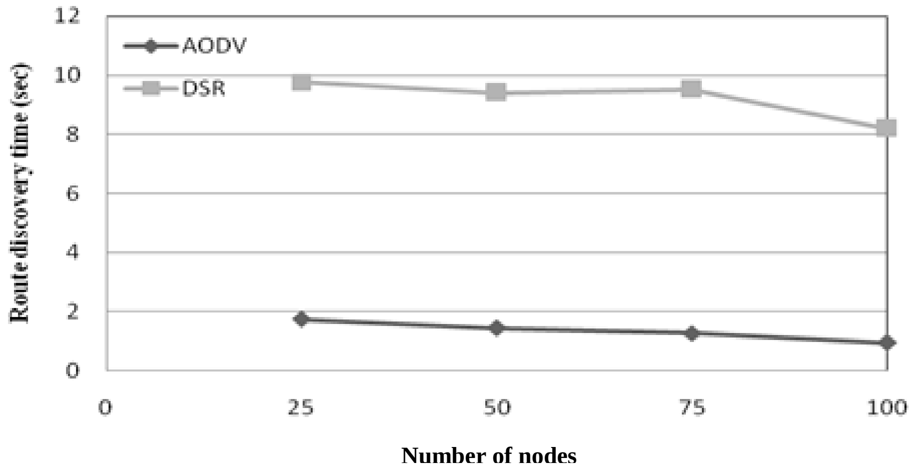


Figure 6.1 Route Discovery Time Versus Number of Nodes for AODV and DSR with one Client using PCM

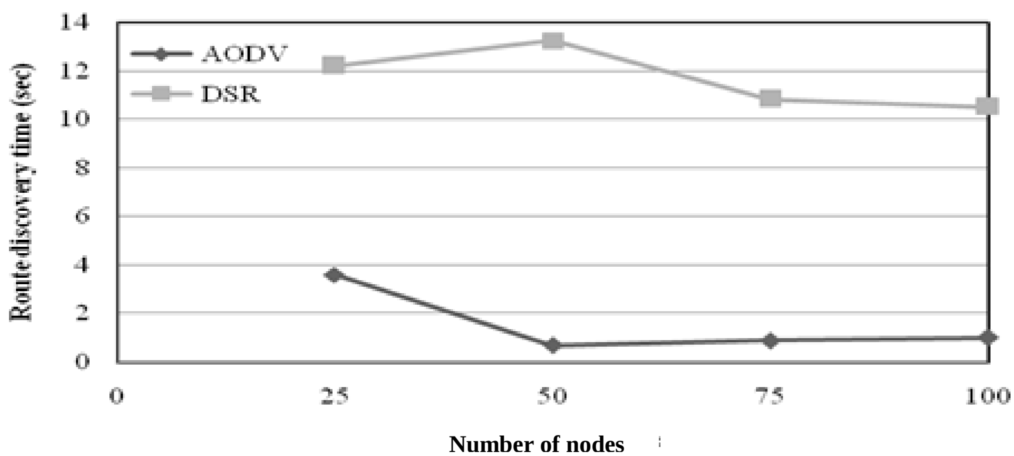


Figure 6.2.Route Discovery Time Versus Number of Nodes for AODV and DSR with 12 Clients using PCM

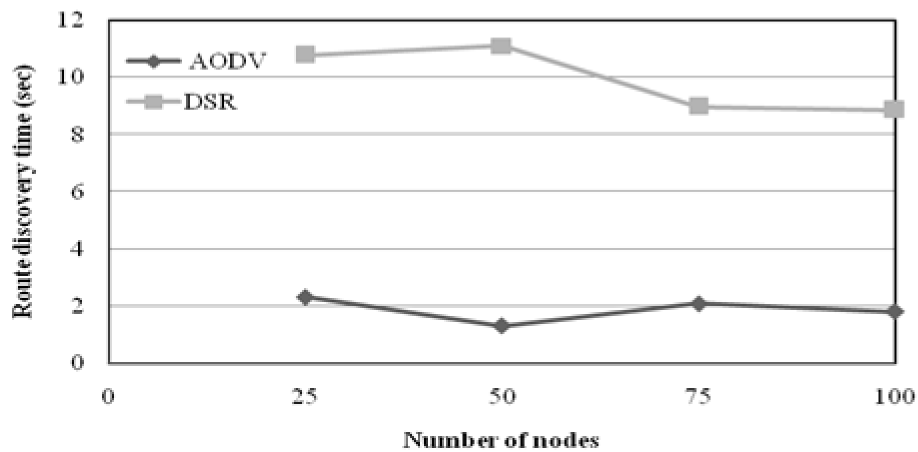


Figure 6.3 Route Discovery Time Versus Number of Nodes for AODV and DSR with One Client using GSM

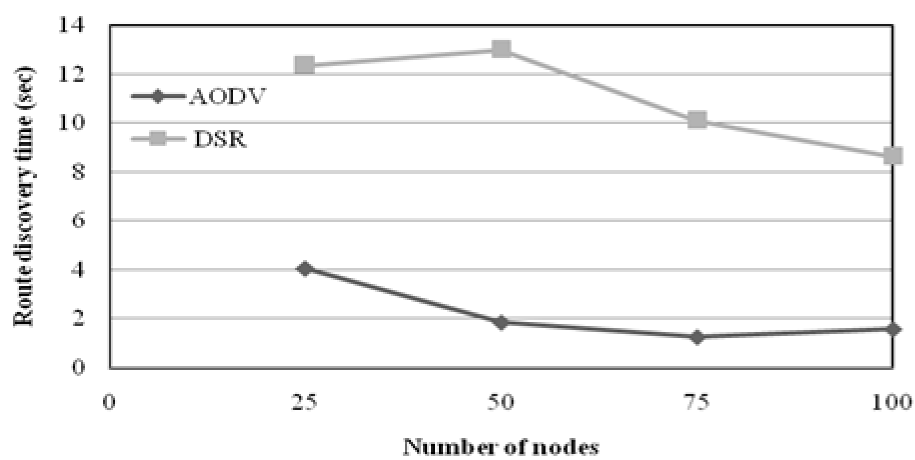


Figure 6.4 Route Discovery Time Versus Number of Nodes for AODV and DSR with 12 Clients using GSM

Figures 6.1- 6.4 show that the route discovery time for DSR is larger than AODV because the route discovery takes a longer time at every intermediate node since DSR tries to extract information before forwarding to the next node. The same thing happens when a data packet is forwarded hop by hop. While the AODV routing protocol makes a route discovery more efficient, and using refreshable and the newest route to the destination. As we increase the number of connected clients, the route discovery time also increases. but it has no effect in the large number of nodes. As the number of node increases, the route discovery time for DSR is decreases. This is due to the large number of alternative multiple routes to the destination node which is cached in its memory during the route discovery time. In additional, the route discovery time for AODV is decreases. This is due to the large number of route request messages forwarded to the destination node with routing information for the newest routes to it. But, AODV still have a less route discovery time compared with DSR protocol. This is because of AODV uses a sequence number to find the newest routes to the destination during the route discovery time.

Figure 6.5 - 6.8 show graphs of the jitter with respect to number of nodes.

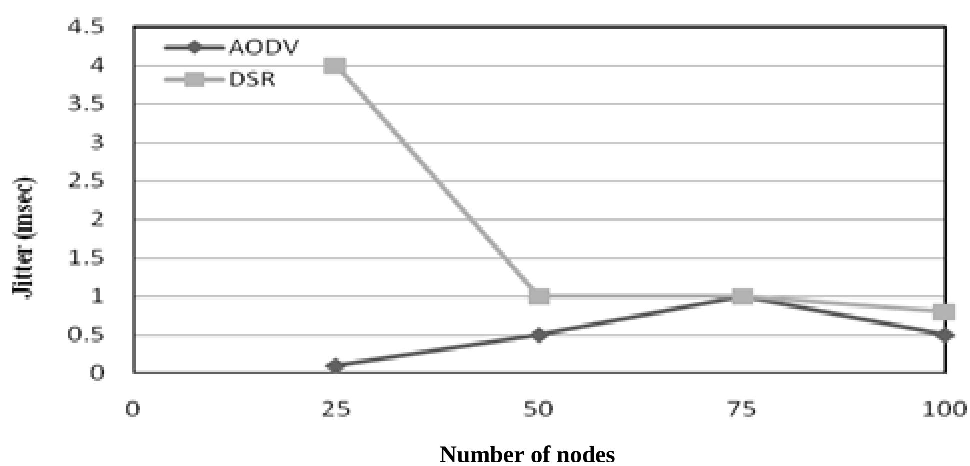


Figure 6.5 Jitter Versus Number of Nodes for AODV and DSR with One Client using PCM

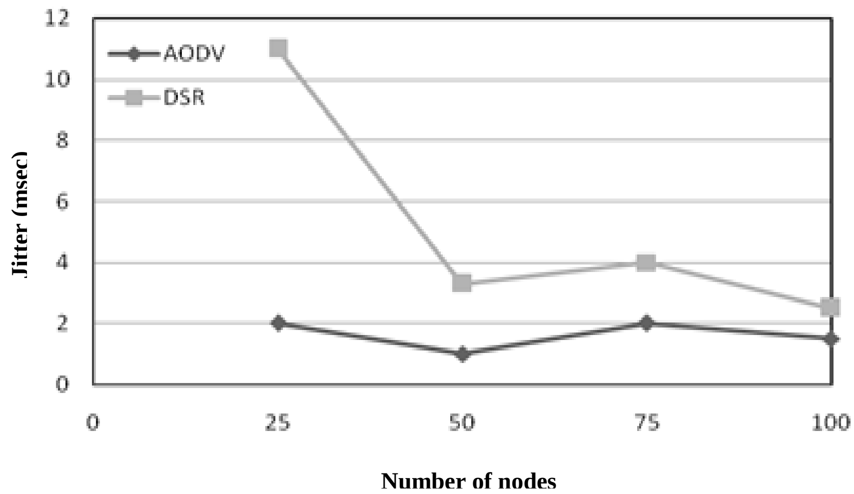


Figure 6.6 Jitter versus Number of Nodes for AODV and DSR with 12 Clients using PCM

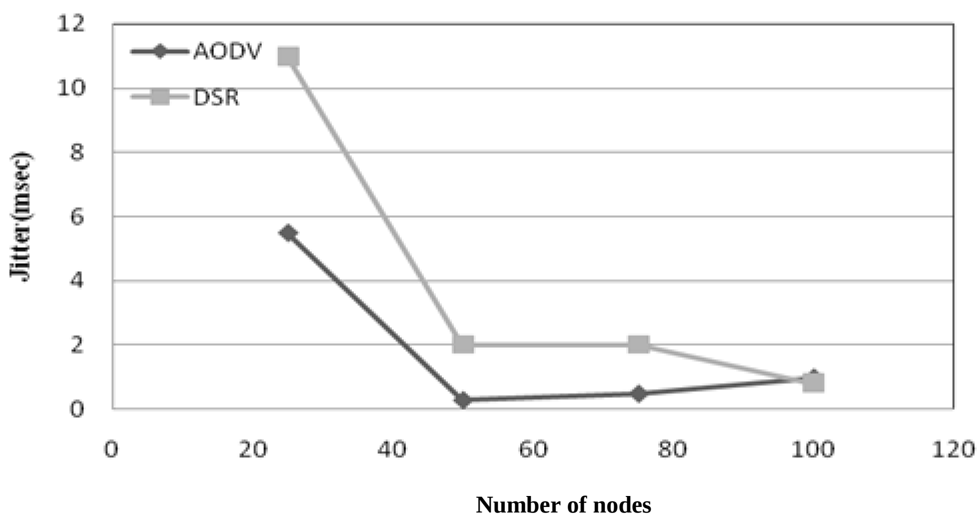


Figure 6.7 Jitter Versus Number of Nodes for AODV and DSR with One Client using GSM

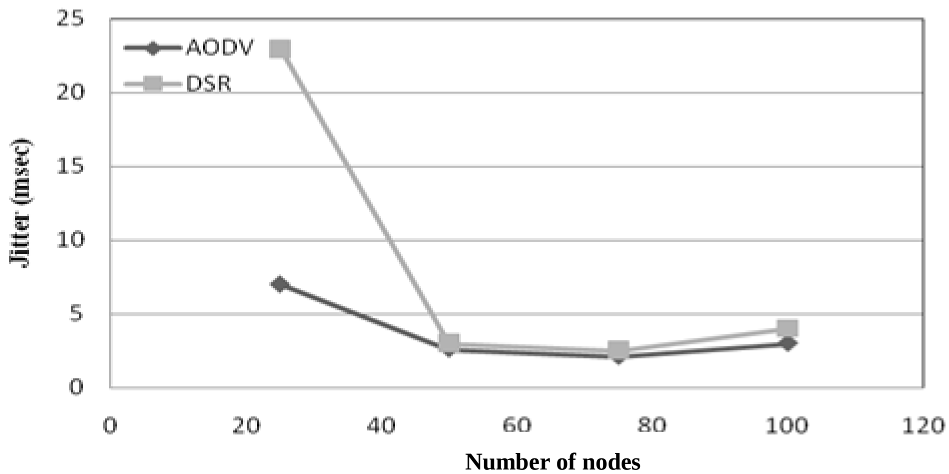


Figure 6.8 Jitter Versus Number Of Nodes for AODV and DSR with 12 clients using GSM

Figures 6.5 - 6.8 show that jitter values for DSR are larger than the jitter value in the AODV MANET routing protocol. As the number of clients increases, the jitter value increases, but it has no effect in the large number of nodes. DSR visit and route through all the possible intermediate nodes to the destination. This increases the jitter. AODV uses the most refrashable and the newest route to the destination. GSM is includes more compression for the voice packet. This affects the quality of voice. GSM has a high jitter compared to PCM. In DSR, there is more probability for jitter as a node broadcasting a route request packet to its entire neighbor nodes in the network. For DSR, increasing the number of nodes causes a decrease in jitter. It is because DSR has multiple routes, during its route discovering process. DSR identifies the multiple routes to the target node which is an increase in high density and low mobility nodes. Therefore, it causes a decrease in route discovery time for the destination node among the intermediate nodes. In additional, the route discovery time for AODV is decreases. This is due to the large number of route request messages forwarded to the destination node with routing information for newest routes to it, during its route discovering process. However, AODV still has less jitter compared with DSR.

Figures 6.9 - 6.12 show graphs of the packet end-to-end delay with respect to number of nodes.

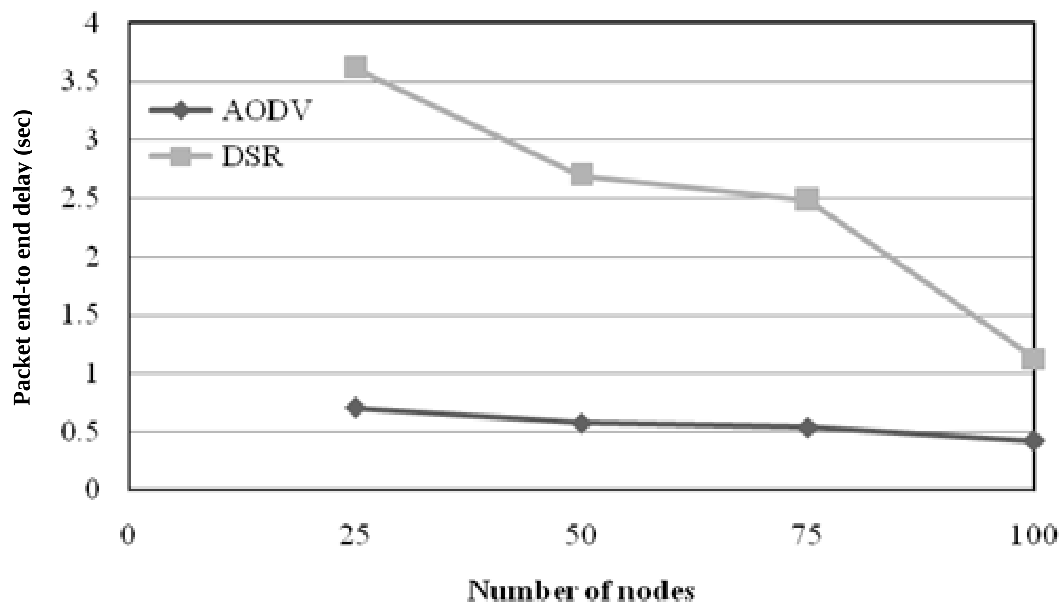


Figure 6.9 Packet End -to-End Delay Versus Number of Nodes for AODV and DSR with One Client using PCM

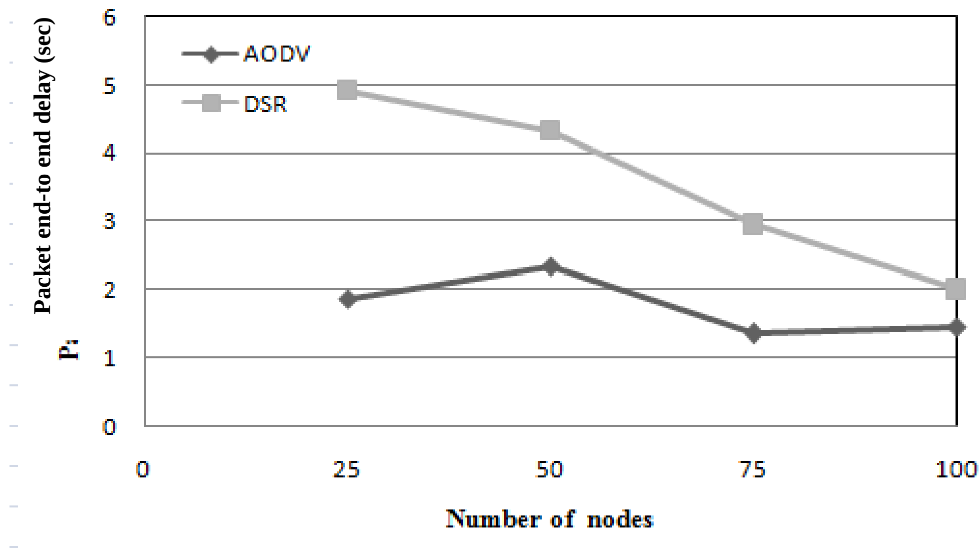


Figure 6.10 Packet End -to-End Delay Versus Number of Nodes for AODV and DSR with 12 Clients using PCM

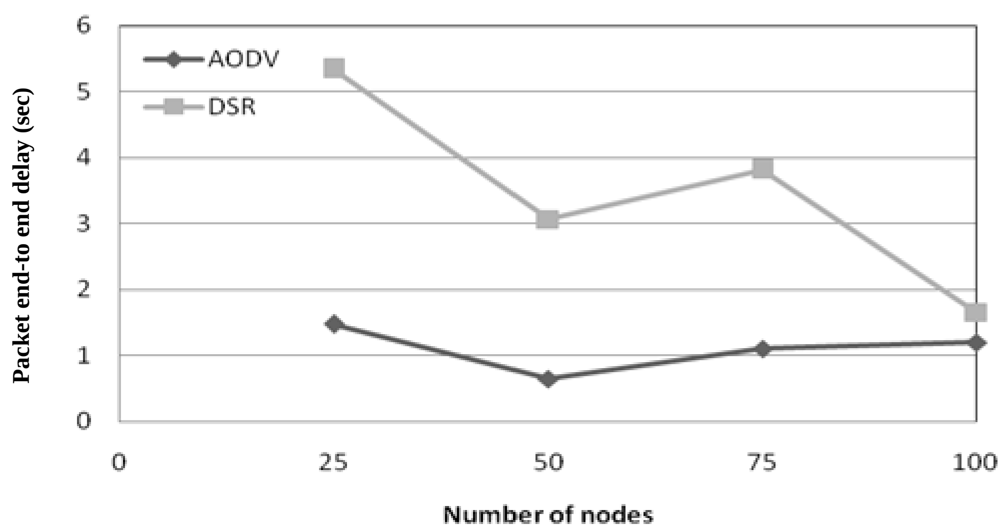


Figure 6.11 Packet End -to-End Delay Versus Number of Nodes for AODV and DSR with one client using GSM

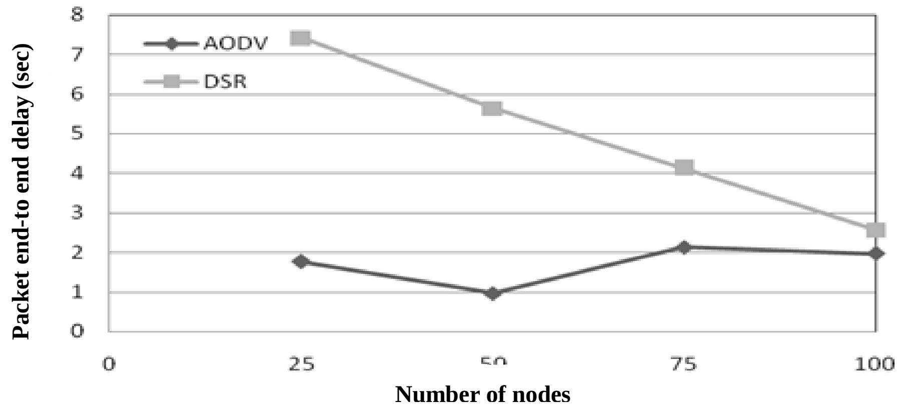


Figure 6.12 Packet End -to-End Delay Versus Number of Nodes for AODV and DSR with 12 Clients using GSM

Figures 6.9 - 6.12 show that packet end-to-end delay for DSR is larger than AODV. This is due to the fact that in the case of congestion or routing overhead, control messages get lost and so decreasing its advantage of fast establishing new routes with DSR routing. Under such conditions, DSR has a relatively high delay. As the number of clients for one server network increases the packet end-to-end delay increases also, but it has no effect in the large number of nodes. For a single client, GSM has more delay over PCM as the number of clients increases. Because of the high compression of GSM, we experience more delay. In DSR, there is a high voice packet end-to-end delay because of an aggressive route caching. Increasing the number of nodes leads to a decreased the delay in DSR. Clearly, large multiple routes to the destination node are increased in high density and low mobility nodes and also it causes a decreasing route discovery time to the destination. Also, the end-to-end delay for AODV is decreases in large numbers of nodes. In additional, the route discovery time for AODV is decreases. This is due to the large number route request messages forwarded to the destination node with routing information for newest routes to it, during its route discovering process.

However, AODV still has a better performance compared with DSR. This is because AODV used the freshness routes to the destination.

Table 6.11 presents the Traffic delivery ratio for one client with AODV and DSR protocols.

Table 6.11: Traffic Delivery Ratio for One Client using PCM/GSM

Number of nodes	**PCM**		**GSM**	
	AODV	**DSR**	**AODV**	**DSR**
25	44%	27%	41%	21%
50	57%	31%	56%	23%
75	55%	53%	52%	36%
100	51%	49%	49%	48%

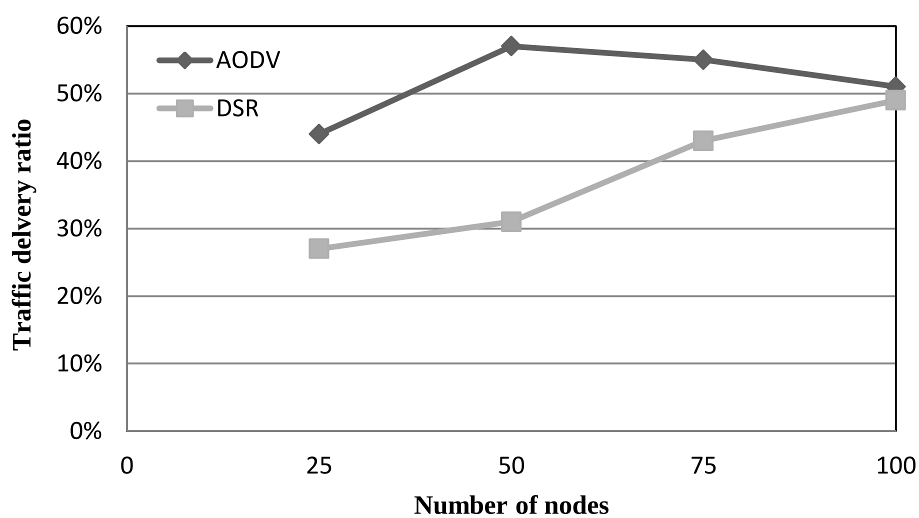


Figure 6.13 Traffic Delivery Ratio Versus Number of Nodes for AODV with using PCM

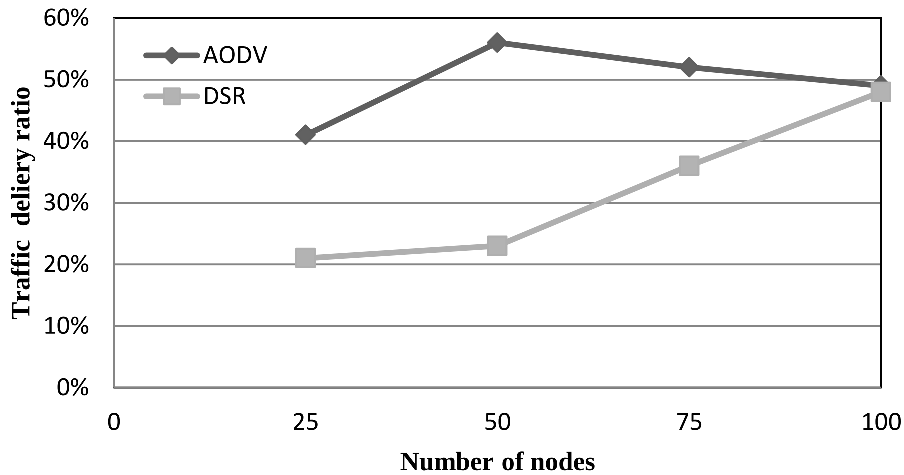


Figure 6.14 Traffic Delivery Ratio Versus Number of Nodes for DSR using GSM

Figures 6.13 and 6.14 shows Traffic delivery ratio with one server and one client. Since PCM has a large bandwidth 64kbps compared with GSM 12 Kbps, the traffic delivery ratio for PCM is larger than GSM. The traffic delivery ratio for AODV is more than for DSR since AODV routes to the most refreshable and it uses route's expiry, dropping some packets when a route expires and a new route must be found. Since DSR has multiple routes, during its route discovering process increasing the number of nodes brings an increase in packet delivery ratio. DSR identifies many routes to the destination node in which these routes are increased in high density and low mobility nodes. Also, the traffic delivery ratio for AODV is decreases in high number of nodes. This is due to the large number route request messages forwarded to the destination node with routing information for newest routes to it, during its route discovering process.

But, AODV has a slightly better more traffic delivery ratio compared with DSR. This is because AODV uses the newest route to the destination compared with DSR.

From the above results, a conclusion was drawn that AODV is better than DSR. GSM presents high delay compared to PCM. Traffic delivery ratio decreases with an increase in the number clients, but it has no effect in the large number of nodes..

Table 6.12 shows summary of our simulation results.

Table 6.12: Summary of Results

Voice schemes	**Number of the clients that are participate with voice application**	**Routing protocol that performs best according to each performances metrics**			
		Route discovery time (sec)	**Jitter (msec)**	**Packet end-to-end delay (sec)**	**Traffic delivery ratio (%)**
PCM	1	AODV	AODV	AODV	AODV
	12	AODV	AODV	AODV	AODV
GSM	1	AODV	DSR	AODV	AODV
	12	AODV	AODV	AODV	AODV

6.2 Comparison with Others Related Work

According to our knowledge, in the related studies, there is no detailed information about the number of clients in the network. Also they used mesh topology for MANET. Almost all performance metrics results were presented with respect to simulation time. So we cannot compare our results with other related work results one to one. Despite of this, we have presented some of our results and other related work results. In Tables 6.13 and 6.14.

Table 6.13: Comparison with [3]

Voice scheme	**PCM**			
source	Ref [3]		Our simulation results	
Routing protocol	AODV	DSR	AODV	DSR
Number of nodes	25	25	25	25
Jitter (sec)	0.00048	0.005	0.0001	0.004

The results of our study are compared with [3] in Table 6.13. It is noticeable that, the used performance metric is jitter here. According to results, one can say, that in [3] the AODV has a low jitter with value 0.00048 sec compared with DSR jitter value 0.005 sec. In our simulation results, the AODV has lower jitter value with 0.0001sec than the DSR jitter value with 0.004 sec.

Table 6.14: Comparison with [2]

Voice schemes	**GSM**			
Source	Ref[2]		Our simulation results	
Routing protocol	AODV	DSR	AODV	DSR
Number of nodes	20	20	25	25
Route discovery time (sec)	4	9	2.305	10.78
Packet end–to-end delay (sec)	3.3	4.5	1.471	5.331

In Table 6.14, we have compared our results with [2]. Performance metrics are used for comparison, route discovery time and packet end-to-end delay. It is observed that in [2], the AODV has lower route discovery time and packet end-to-end delay with values 4 sec and 3.3 sec respectively compared with DSR by having values 9 sec and 4.5 sec. In our simulation results, the AODV discovery time and packet end-to-end values (2.3 sec and 1.4sec) respectively are lower than DSR values (10.7sec and 5.33sec).

6.3 Confidence Interval Calculation

In this thesis, to calculate the confidence interval the replication method was used. More than one runs are used with different seed values. Different seed values generated different random numbers for the same simulation time for each run. After that the standard deviation was calculated using the expression below:

$$Confidence\ Interval\ (CI) = \bar{x} \pm t^* \frac{s}{\sqrt{n}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ 6.1$$

$\bar{X} =$ *mean*

S = *standard deviation*

n = *number of runs*

t^* = *critical value*

Where Standard deviation is calculated as :

$$S = \sqrt{(x_1 - \bar{x})^2 + (x_2 - \bar{x})^2 + (x_3 - \bar{x})^2 + \cdots} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ 6.2$$

Here $x_{i,\ i=1\ to\ n}$ *is the result of each run and mean is the average of n runs as shown below:*

$$\bar{x} = \frac{x_1 + x_2 + x_3 + \cdots x_n}{n} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ 6.3$$

t^* *is calculated using formula TINV (1-level, n-1)*

Level = confidence interval or confidence level

n-1= The degree of freedom.

Using confidence interval equation two values are found, which present the confidence range measurement. Tables 6.15 - 6.18 present the confidence interval results with 1 client in the network

Table 6.15: Confidence Interval for DSR Routing Protocol PCM Voice Scheme for One client

Performance Metric	**Number of nodes**			
	25	50	75	100
Route discovery time (Sec)	1.921∓1.435	1.851∓1.074	1.851∓1.117	1.701∓ 0.715
Jitter (Sec)	0.0001∓ 0.00013	0.0005∓ 0.00041	0.001∓ 0.001806	0.0005∓ 0.0003
Packet end to end delay (Sec)	0.704∓0.615	0.572∓0.566	0.539∓0.388	0.417∓ 0.400
Traffic receive (packet/Sec)	59.721∓10.164	82.76∓22.931	77.325∓21.039	72.053∓ 38.648
Traffic sent (packet/Sec)	133.390∓ 10.164	144.707∓6.594	141.991∓6.800	139.622∓ 15.21

Table 6.16: Confidence Interval for AODV Routing Protocol PCM Voice Scheme For one Client Node.

Performance Metric	**Number of nodes**			
	25	50	75	100
Route discovery time (Sec)	11.759∓3.910	12.400∓0.864	11.685∓1.125	11.893∓0.654
Jitter (Sec)	0.004∓*0.003*	0.001∓0.0008	0.001∓0.00	0.0008∓0.0007
Packet end to end delay (Sec)	3.610∓1.891	2.688∓2.298	2.484∓0.398	1.124∓0.699
Traffic receive (packet/Sec)	36.83∓32.233	38.034∓14.851	73.279∓32.935	70.568∓14.472
Traffic sent (packet/Sec)	132.817∓7.763	123.055∓ 5.980	138.221∓17.643	136.644∓ 4.946

Table 6.17: Confidence Interval for AODV Routing Protocol GSM Voice Scheme for One Client Node

Performance metric	**Number of nodes**			
	25	50	75	100
Route discovery time (sec)	2.305∓0.310	1.299∓0.690	2.074∓0.933	1.794∓0.867
Jitter (sec)	0.055∓0.016	0.0003∓0.00019	0.0005∓0.00048	0.015∓0.0009
Packet end to end delay (sec)	1.471∓0.497	0.645∓0.043	1.106∓1.111	1.196∓0.754
Traffic receive (packet/sec)	141.852∓ 36.016	202.121∓52.108	186.471∓47.169	175.657∓100.49
Traffic sent(packet/sec)	339.502∓ 14.500	360.274∓31.406	356.169∓18.366	354.488 ∓39.843

Table 6.18: Confidence Interval DSR Routing Protocol GSM Voice Scheme for One Client Node

Performance metric	**Number of node**			
	25	50	75	100
Route discovery time (Sec)	10.781 $\mp$ 0.559	11.107 $\mp$ 1.491	8.971 $\mp$ 1.872	8.849 $\mp$ 0.747
Jitter (Sec)	0.011 $\mp$ 0.0094	0.002 $\mp$ 0.001	0.002 $\mp$ 0.0021	0.0008 $\mp$ 0.00067
Packet end to end delay (Sec)	5.339 $\mp$ 1.424	3.065 $\mp$ 2.851	3.821 $\mp$ 3.734	1.654 $\mp$ 1.611
Traffic receive (packet/Sec)	12.512 $\mp$ 4.306	71.141 $\mp$ 42.478	123.835 $\mp$ 104.158	167.403 $\mp$ 130.463
Traffic sent (packet/Sec)	337.067 $\mp$ 18.692	298.376 $\mp$ 21.201	336.709 $\mp$ 35.843	342.421 $\mp$ 52.234

Chapter 7

CONCLUSION

In this thesis, a study was conducted for ad hoc wireless network environments with voice applications. OPNET 17.1 was used in the simulation of the thesis to model an ad hoc with voice application. AODV and DSR routing protocols were analyzed with PCM and GSM voice schemes using different number of the clients (1,12) in varying number of nodes (25, 50, 75 and 100) in the MANET.

Two classes of performance metrics were used in the evaluation of the system. The first class of performance metrics was for MANET routing (route discovery time). The second class was for voice application (jitter, packet end-to-end delay, traffic sent and traffic receive).

In this thesis, a simulation study for MANETs was conducted with different network parameters. In all cases, there is only one server and numbers of client were set to 1 and 12 for different number of nodes of MANET. Nodes were distributed randomly in the network and random waypoint mobility was used for mobility of nodes.

From the extensive simulations, we can conclude that AODV performance is better than DSR. AODV maintains only one entry per destination in the routing table. The destination replies only to the first arrived request and ignores the rest. While DSR uses route caching aggressively and replies to all requests reaching a destination from a single request cycle.

Simulation results also show that AODV reactive routing protocol is the best suited for MANET networks with GSM and PCM voice encoding scheme compared with DSR. But, DSR has multiple routes, during its route discovering process increasing the number of nodes make DSR identifies many routes to the destination node in which these routes are increasing in high density and low mobility nodes,that causes a decrease in route discovery time for the destination and make DSR better. But, AODV is a slightly better compared with DSR. This is because AODV uses the newest route to the destination compared with DSR.

Simulation results show that PCM has low end-to-end delay compared to GSM. Traffic delivery ratio of voice packets in PCM encoding schemes is high.

Jitter is high in GSM because of the compression, decompression, encoding and decoding operations. Since large frame sizes require more time for compression and decompression. This has a great effect on the quality. However, GSM doesn’t require more bandwidth only 12.5Kbps. PCM has more quality and doesn’t include any compression delay, but it requires more bandwidth (64Kbps).

Finally, PCM has a good quality and low end-to-end delay, but it requires a high bandwidth. GSM has fair quality and high delay but doesn’t consume bandwidth. In addition, it was determined that AODV is best to use with both voice schemes PCM and GSM with a different number of clients and one server.

In general, one can say that with low node speed (walking speed) AODV and DSR have almost the same behavior in a large node density. But still AODV has better performance than DSR.

For future work, simulation for more than two MANET routing protocols with different mobility speeds might be constructed by increasing the number of clients and including more testing performance metrics.

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