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Titre de la Thèse **: Routage Adaptatif pour la Gestion de la Qualité de Service : Cas des Applications VoIP dans les Réseaux Mobiles Ad Hoc (VoMAN)**

Soutenue le 26 Octobre 2013 devant le Jury

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TITLE

Adaptive Routing for Quality of Service Management: Case of VoIP Applications over Mobile Ad hoc Networks (VoMANs)

Said EL BRAK

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To my Parents

To my Wife

And

Dearest Mohamed Amine.

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Abstract

Nowadays, Mobile Ad hoc Networks (MANETs) have to support new applications including VoIP (Voice over IP) that impose stringent QoS constraints and high delivery requirements. However, VoIP applications make a very inefficient use of the MANET resources due to several factors such as: large overhead caused by frequently transmitting small packets, signalling overhead associated with routing protocols, and route selection based on shortest path without taking into consideration other conditions. These factors make bandwidth increasingly limited, and thus, the voice quality will be degraded.

Our work represents a first step toward improving aspects at the network layer by addressing issues from the standpoint of adaptation, claiming that effective adaptation of routing parameters can enhance VoIP quality. Firstly, we established a state of art about the technical building blocks of the subject. Then, we have performed extensive simulation studies which have been highly beneficial to define parameters that need adaptation. We studied the impact of tuning OLSR routing protocol parameters on VoMAN. This research has helped us to understand the behaviour of VoIP codecs when varying OLSR parameter values. We showed that a quantitative relationship between VoIP codecs performance and factors like refresh intervals and willingness; hence, an adaptation strategy and protocol is needed.

The most important contribution of the thesis is the adaptive OLSR-VA algorithm which provides an integrated environment where VoIP activity is constantly detected and routing parameters are adapted in order to meet the application requirements while minimizing the induced overhead. The proposed routing adaptation algorithm is composed by two phases: Monitoring and Adaptation. We have extended the OLSR protocol by including adaptation functions and modifying heuristic routing. Furthermore, new HELLO message structure has been used for the dissemination of the VoIP activity information. To investigate the performance advantage achieved by such algorithm, a number of realistic simulations (VANET and healthcare application) are performed under different conditions. The most important observation is that performance is satisfactory, in terms of the perceived voice quality and call capacity. Results have shown ability of the solution for successfully achieve an acceptable voice quality even over long routes and under reasonably load conditions. The proposed scheme can be integrated into a generic QoS adaptation management architecture.

Key words: MANETs, VoIP, QoS Management, codec, OLSR, ns-2 c

Résumé

Les réseaux mobiles ad hoc ou MANETs (Mobile Ad hoc NETworks) sont des réseaux formés spontanément par des terminaux mobiles communiquant directement entre eux via le medium radio. Ces nœuds mobiles n'ont plus recours à une infrastructure de communication fixe ou préexistante. Vu que tel réseau peut être déployé n'importe où et n'importe quand, la Voix sur IP (VoIP) à travers les réseaux MANET (VoMAN) est considérée comme une application ambitieuse. Cependant, le VoIP étant l'une des applications de communications qui exige des besoins stricts en terme de qualité de service (band passante, délai, gigue, et taux de perte), les réseaux MANET, quant à eux, souffrent de plusieurs contraintes et doivent ainsi évoluer techniquement et économiquement pour rendre cette application acceptable. Dans un tel contexte, les VoMANs ne pourront s'imposer que dans la mesure où la qualité de la parole reçue sera satisfaisante.

Dans la présente thèse, notre solution est basée sur la gestion de QoS en impliquant une approche adaptative. L'idée d'adapter les paramètres de protocole de routage aux caractéristiques de l'application VoIP est le point de départ de notre recherche. En premier lieu, nous avons effectué une étude de simulation approfondie qui porte sur trois volets : le premier consiste à confirmer l'efficacité du protocole de routage proactive OLSR (Optimized Link State Routing) pour les applications VoIP, le second volet, permet de tester la performance des codecs audio dans l'environnement MANET, alors que le troisième relève une étude sur l'impact de la mise au point des paramètres de routage sur la performance de VoMAN. Cette étude nous a permis de confirmer l'existence d'une relation entre les paramètres de routage et l'activité VoIP dans le réseau. Lors de la deuxième contribution, nous nous sommes intéressés à la conception d'un mécanisme de routage adaptatif en intégrant des fonctions d'adaptation au protocole OLSR, de telle manière à ce que ce protocole de routage soit conscient aux activités VoIP et du codec audio utilisé pendant les communications vocales. Dès lors, notre extension est nommée OLSR-VA (OLSR-VoIP-Aware). Nous avons évalué par simulation les performances de notre solution en considérant des scenarios réels. Le premier concerne la transmission de la voix dans le réseau véhiculaire (VANET) et le deuxième implémente VoMAN dans une application de soins de santé (Healthcare). Notre objectif était de mesurer les métriques QoS d'une façon quantitative et qualitative. Les résultats menés par notre solution ont montré sa capacité pour atteindre une qualité acceptable dans des conditions de charge raisonnable.

Mots clés : Réseau mobile ad hoc, Voix sur IP, Qualité de service, Architecture de gestion, Routage OLSR, simulation ns-2.

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List of Acronyms

ACK	Acknowledgement
AMR	Adaptive Multi-Rate
ANMP	Ad Hoc Network Management Protocol
AODV	Ad hoc On Demand Distance Vector
AP	Access Point
APP	Application layer
BSN	Body sensor Network
CAC	Call Admission Control
CELP	Code-excited linear prediction
Codec	Coder Decoder
COPS	Common Open Policy Service
CSMA/CA	Carrier Sense Multiple Access / Collision Detection
CTS	Clear To Send
DARPA	Defense Advanced Research Projects Agency
DCF	Distributed Coordination Function
DREAM	Distance Routing Efect Algo- n.thm for Mobility
DSDV	Destination-Sequenced Distance Vector
DSR	Dynamic Source Routing
EDCA	Enhanced distributed channel access
GPSR	Greedy Perimeter Stateless Routing
GSM	Global system mobile
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU-T	International Telecommunication Union-Telecommunication
LLC	Link Layer Control
LVA	Local VoIP Activity
MAC	Media Access Control
MANET	Mobile Ad hoc Network
MIB	Management Information Base
MOS	Mean Opinion Score
MPR	Multi-Point Relay
NAV	Network Allocation Vector
NET	Network layer
NRL	Normalized Routing Load
NS	Network Simulator
NVA	Neighbour VoIP Activity

OAR	Opportunistic Auto Rate
OLSR	Optimized Link State Routing Protocol
OLSR-VA	Optimized Link State Routing Protocol- VoIP Aware
PBNM	Policy-based Network Management
PCF	Point Coordination Function
PCIM	Policy Core Information Model
PCM	Pulse Code Modulation
PDP	Policy Decision Point
PEP	Policy Enforcement Point
PHY	Physical layer
PR	Policy repository
QDVP	Quality degradation of VoIP Packets
QoS	Quality of Service
RFC	Requests For Comments
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
RTS	Read To Send
RWP	Random Way Point
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol
SPF	Shortest Path First
TBRPF	Topology-Based Reverse Path Forwarding
TC	Topology Control
UDP	User Datagram Protocol
UP	User Priority
VAD	Voice Activity Detection
VANET	Vehicular Ad hoc Network
VoIP	Voice over IP
VoMAN	VoIP over MANET
WLAN	Wireless Local Area Network
WMN	Wireless Mesh Network
WPAN	Wireless Personal Area Network
WSN	Wireless Sensor Network
ZRP	Zone Routing Protocol

Chapter 1

Introduction

1.1. Motivation

The overall rapid growth of wireless technologies has changed our lives in all kinds of useful ways. Many services are moving from household based to individual based services. For instance, voice communications have been transitioned from fixed line phones to mobile phones, to texting, and video calls over the all-IP networks. Moreover, another evolution of wireless technologies is providing mobility support for users which open the way for large wireless mobile network applications and standards, ranging from large-scale networks (3G/4G access networks) to small-scale networks (WLAN, WPAN, etc.).

Another emerging wireless technology is the *Mobile Ad hoc Networks* (MANETs), which are ubiquitous networks, can be flexibly and conveniently deployed in almost any environment. MANETs are self-forming and self-healing, enabling peer-level communications between mobile nodes without relying on centralized resources or fixed infrastructure. Recently, these networks are gaining more and more popularity and receiving tremendous attention from the research community. Moreover, MANETs have been the originator platform for other networking areas as VANETs, WSNs, BSN, and WMNs.

Voice over IP (VoIP) is one of the fastest growing applications in networking. As wireless components spread, VoIP over wireless is becoming increasingly important. Hence, as part of a greater IP network, a MANET is expected to support real-time traffic and, in particular, voice. Thus, supporting voice over ad hoc networks is part of realizing an all-IP goal.

Figure 1.1: Emergency inter-vehicle voice communications scenario

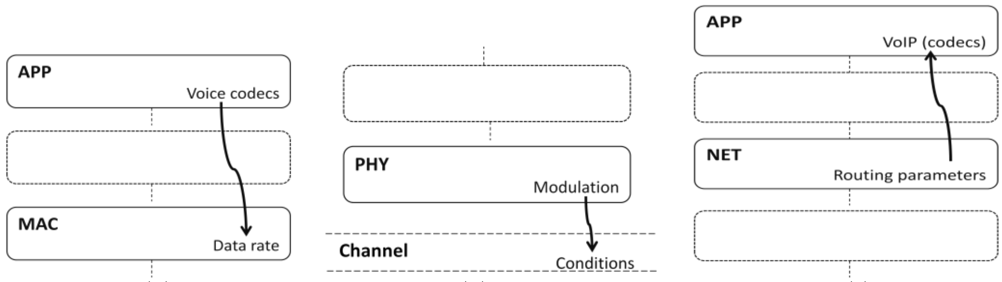
MANET's attributes enable to provide VoIP services in virtually any scenario key applications including: disaster recovery, heavy construction, mining, transportation, defense, and special event management. In the following, we detail some examples of *VoIP over MANET application* (VoMAN) scenarios:

Handoff: MANETs might corporate as extensions with infrastructure-based networks (cellular networks and wireless LANs) for maintaining voice communications in *dead zones*: places where the access points cannot reach. In [80], Ad Hoc Assisted Handoff is proposed wherein nodes that are outside the coverage of any *Access Point* (AP), yet are in the vicinity of one, use nodes within the range of the AP as forwarding nodes.

Emergency: When accidents occur in hard-to-reach locations such as in a tunnel or impassable mountain terrain where no infrastructure networks are found. MANETs could enable rescue workers to communicate with one another or with a command control centre [81]. Figure 1.1 illustrates a scenario where emergency vehicle (e.g. police cars, ambulances, fire trucks, etc.) need voice communication in a disaster struck areas which lack telecommunication infrastructure.

Security: Other scenarios include surveillance systems that detect an intruder, habitat monitoring or monitoring in a biosensor network. In these scenarios the transmission of voice over wireless sensor networks is needed [82-83].

In this work, we consider only wireless networks capable of operating without the support of any fixed infrastructure. We also consider the general case of multi-hop networks. More precisely, we assume a best-effort network that carries data and voice traffic without differentiation. Additionally, our work focuses on VoIP traffic transmission; we do not address signalling issues.



(a) (b) (c)

Figure 1.2: Adaptation solutions

1.2. Problem statement

Naturally, users demand to be able to use the same VoIP services independently of the access network. However, many QoS (Quality of Service) issues remain unsolved in infrastructure-based networks [20]. So how it will be for the MANETs which combine many challenges? At first, the wireless channel introduces constraints including its inherent broadcast nature and temporal response variability due to fading, absorption and noise and interference-sensitivity. In addition, ad hoc networks suffer from scarcity of resources, lack of a central entity, and volatility of connections. These challenges create new performance limitations for VoMAN and make it a new exiting task.

Several works have been devoted to evaluate voice quality transmission over MANET [42-44-45-46-84-68-85-86-87-88]. Among of these, works have been shown that VoIP applications make a very inefficient use of the MANET resources due to several factors such as: large overhead caused by frequently transmitting small packets, signalling overhead associated with routing protocols, and route selection based on shortest path without taking into consideration other conditions. These factors make bandwidth increasingly limited, and thus become voice quality degraded.

The first QoS solutions for VoIP over MANETs have addressed traditional QoS guaranteeing mechanisms already designed for WLAN, such as: resource reservation [89], admission control techniques [90-91], and differentiated services (802.11e standard) [93-92] which can guarantee some minimum QoS for multimedia traffic. Also, the link and codec adaptation mechanisms [78-79] were one step ahead to construct QoS solution for VoIP over multi-rate wireless link. These mechanisms are providing resilience and robustness in cellular networks [94] hence, is required to provide MANETs the same capabilities.

Adaptation mechanisms can be done at several layers of the network protocol stack. Well known examples of adaptation mechanism for VoIP over wireless media are: PHY/Channel adaptation which allows each mobile node to adapt their transmission rate dynamically to channel condition [95] (Figure 1.2(a)). APP/MAC adaptation adapts the codecs of the active voice flows to the new network conditions (data rate) (Figure 1.2 (b)). The last one is considered one of the most effective solutions of the multi-rate problem on VoIP calls [78-79].

Routing layer is one of the major key components for the MANET connectivity and performance. Nevertheless, few efforts have focused on the impact of routing protocols on voice traffic over MANETs. Also, adaptation mechanisms have to deal with routing plan and integrate routing protocol in their architecture. Based on these considerations, and to further improve the performance of voice over MANETs, we propose an adaptation mechanism that integrates routing in the solution. My thesis is to make routing protocol aware of the voice travel and improve the quality of the voice over MANETs. I propose a self-adapting routing protocol that adapts it parameters to the VoIP load in the network and to voice codec setting. To the best of our knowledge, adapting the routing parameters to media transmission has never been addressed before. The next section describes in detail my contributions.

1.3. Contributions

My work focuses on investigating self-adaptation mechanisms, since MANETs are autonomous networks. The self-adaptation means that several network protocols parameters should be automatically configured and adapted to guarantee acceptable QoS [96]. As mentioned in Figure 1.2 (c), my proposal for quality of service in VoIP over MANET has focused on adaptation between network layer and application layer by integrating self-adaptation mechanism in the routing protocol. The proposed adaptation approach may be implemented complementarily with the other adaptation solutions. Before addressing the solution design and implementation, I have done a comprehensive simulation studies which allows us to specify the problem situation.

1.3.1. Comprehensive studies

As a first step, I perform a comprehensive study which lies in performance evaluation of the VoIP application in both reactive and proactive routing through metrics such as: packet loss, delay and jitter, which are the most common ones used for QoS evaluation. Then, I have tested the behaviour of VoIP codecs in order to assess them under static and dynamic MANET environment. Finally, we have identified the impact of routing parameters on VoIP traffic, and from this, select the main parameters whose adaptation appears important to the voice quality experienced. The study shows a dependency between the codec setting (sample size, sample interval) used for voice communication and the routing protocol parameters. I opted the well known ns-2 simulator [99] in my simulation-based studies. The results are presented in Chapter 3.

1.3.2. Proposed solution

I have elaborated my proposals on the basis of the extensive simulation studies performed in Chapter 3. Taking OLSR [28] as the routing protocol baseline, I'm investigating a new integrated application/routing adaptation policy based on a routing metric related to both the VoIP activity and audio codec information. My OLSR extension is called OLSR-VA (OLSR-VoIP-Aware) and is considered the main contribution of this work. This nomenclature is given because OLSR protocol will be aware of VoIP load in the network and the nature of codecs used in VoIP communications. How can this be done?

Firstly, by exploiting layer triggers mechanism; OLSR-VA takes in consideration if a VoIP session is activated in the local node or in it neighbour nodes. A metric has been introduced which is a Boolean value indicates if a VoIP activity is triggered or not. This metric will help in adapting a routing parameter which is the willingness. Based on this, heuristic routing is changed as follow:

- Nodes which are involved in a VoIP activity will not accept to act as MPR (Multi-point Rely) and this leads to preservation of node resources.
- VoIP packet will be routed in paths presenting less VoIP load, and this leads to network bandwidth optimisation.

Secondly, OLSR-VA adapts other parameter (which is the refresh interval) according to codec used in the VoIP communications. The signalling process of OLSR adds a significant overhead to the network by broadcasting control packets in specific intervals, based on this:

- Refresh interval will be reduced if the codec generates VoIP packet with high sample size. As a result, the routing protocol preserves bandwidth which is very required by the above-mentioned codec.
- On the other hand, codecs with small sampling interval generate voice frames in shorter time interval. So, high overhead is required to identify an appropriate path from the sender to the receiver.

Finally, I evaluate the performance of my solution considering VoIP application over VANET in urban environment scenario using quantitative and qualitative metrics. Additionally, a healthcare application scenario was also investigated in order to evaluate the call capacity. Simulation results have shown that the proposed scheme gives better performance, compared to traditional approaches, by selecting paths with high bit-rate links, while also avoiding areas of MAC congestion. Additionally, based on investigating various scenarios, our solution OLSR-VA perform better than OLSR to deal with codecs change, and provide acceptable call quality and capacity.

Additionally, we introduce an adaptive autonomous management system for VoMAN to allow mobile VoIP nodes to configure automatically their protocols parameters and to self-adapt to environment change. However, this architecture needs further investigation and implementation in order to test it performance. This may be a matter for future work.

1.3.3. Methodology

Testing MANET applications before their deployment is an important step in protocols development. Given the cost of implementing such algorithm in real networks, simulations provide a good cost effective environment to study its feasibility and scalability. Additionally, simulation based methodology has persisted for over thirty years which confirm its effectiveness in modelling situation and problems.

In this study, the discrete event network simulator ns-2 [99] was used combined with different frameworks and modules, aiming at a significant level of simulation accuracy. Table 1.1 summarizes tools used in this thesis. Ns-2 is the main simulation tool chosen because it presents many features making it the favourite candidate for implementing and testing my solution. In addition to ns-2, other complementary modules and frameworks are used in order to integrate protocols missing in the original release (e.g. OLSR, VoIP application, etc.). Other tools are used for pre-simulation (i.e. mobility generation) or for post-simulation (i.e. data gathering and processing).

Simulation methodology is based on two types of simulation scenarios (*synthetic scenarios* and *concrete scenarios*). For each scenario, we generate different instances having the same simulation parameters and characteristics. Only simulation seeds which are varied each time generating different instances. We will find more details about our experimental methodology in Section 3.2.

Tool	Scope
Ns-2 [99]	Main network simulator
UM-OLSR [107]	OLSR implementation for ns-2
Ns2voip++ [104]	VoIP traffic simulation for ns-2
IVTG [115]	Vehicular Traffic Generator
OpenStreetMap [116]	Web mapping service
Ns2measure [105]	VoIP measures collection
AWK programming	ns-2 trace files processing
Shell scripting	Automate the simulation process
Gnuplot/Excel	Building graphs

Table 1.1: Tools used in this thesis

1.4. Thesis organization and scope

After introducing the problem and the objectives in this chapter, the remainder of the dissertation is organized as follows:

Chapter 2 provides background information about major keys of this thesis which are: VoIP, MANET and QoS. Firstly, we review the typical VoIP system's elements. Secondly, background on the MANETs is given with an emphasis on the proactive signalling mechanisms. Thirdly, we discuss QoS issues in MANET by reviewing QoS network management approaches designed for MANETs. Afterward, we address the VoIP over MANET, as also we present a review of literature.

Simulation provides an attractive method for evaluating the performance. Chapter 3 is complete and comprehensive simulation studies of VoIP behaviour over MANET. In first, we present the useful guidelines for our experimental design. Then, the performance of VoIP codecs over MANETs is evaluated. Finally, the impact of tuning routing parameters is investigated in order to choose the appropriate adaptation parameters.

Based on investigation performed in Chapter 3, in Chapter 4, we design an adaptation architecture based routing, integrate it within a VoIP over application/network stack, and study its performance over multi-hop networks. Additionally, we briefly introduce a generic adaptation architecture for VoMAN.

Finally in Chapter 5, the main conclusions of this research study are outlined and some future work guidelines are given. The following Figure 1.3 illustrates our system architecture study which entails four major stages.

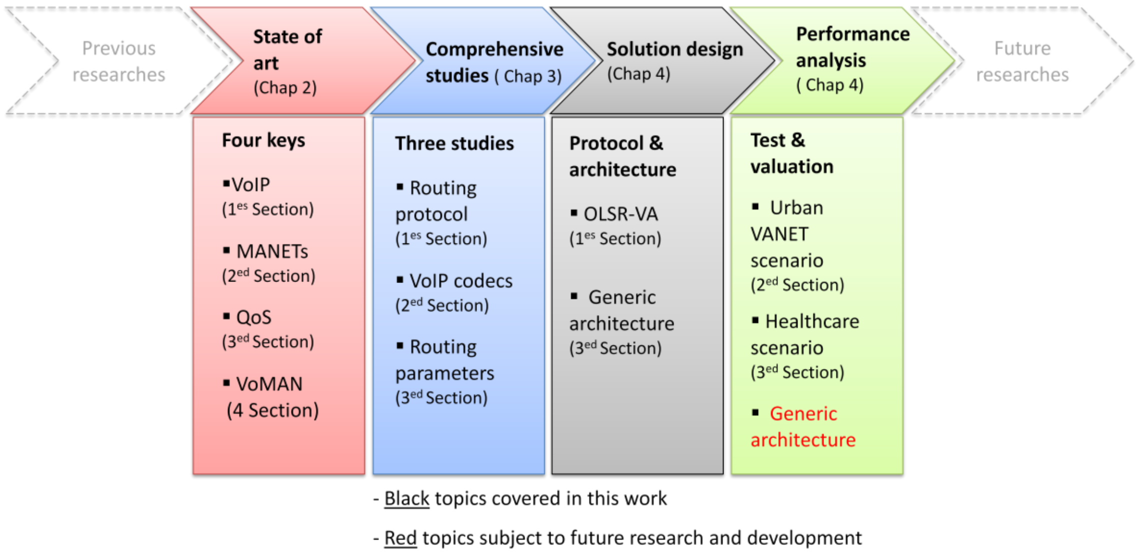


Figure 1.3: Scope of the study

Chapter 2

Background & Literature review

2.1. Introduction

In order to understand the remainder of this thesis, this chapter provides the required background information related to main building blocks of this dissertation. Figure 2.1 illustrates these architectural blocks, including VoIP, MANET, QoS, and their interactions, including VoMAN.

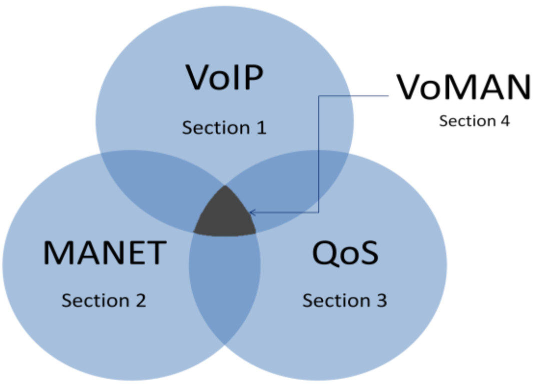


Figure 2.1: Main building blocks of this work

Firstly, we present basic concepts related to VoIP system, VoIP protocols, voice codecs, and voice QoS evaluation methods. Then, we introduce an overview of mobile ad hoc networks and present their general aspects such as routing which has gained a very important interest by research community. Next, we illustrate the major concerns of some well-known QoS management approaches for mobile ad hoc networks.

Due to the ease-of-use of mobile ad hoc networks, voice transmissions over such networks become a hot research topic. The main intention in this chapter is to provide a high-level overview of some issues surrounding voice transmission over MANETs. Hence, in the last Section, we present work related to voice support over MANETs. Additionally, we emphasize on the aspects of solutions dedicated to QoS enhancement in VoMAN.

2.2. Voice over IP: Overview

The telephone system which is commonly known as the *Public Switched Telephone Network* (PSTN) has been evolving ever since Alexander Graham Bell made the first voice transmission over wire in 1876. Recently, the new all-IP operating realities increasingly dominate, so as it is becoming the infrastructure for phone call as well, which favoured the emergence of new technology called IP-telephony or *Voice over IP* (VoIP).

VoIP commonly refers to the technologies, communication protocols, and real-time transmission techniques involved in the delivery of voice communications over Internet or a private data network. In simple terms, Instead of using conventional landlines, people can make phone calls via the IP networks [1].

The telephone and the Internet were the foundation blocks that VoIP was built upon. It was the accumulation of 40+ years of work in both the telecommunications industry and the computing industry that allowed the existence of VoIP. InternetPhone was the first VoIP product of small company called Vocaltec, started in February of 1995. By 1998 some entrepreneurs started to market PC-to-phone and phone-to-phone VoIP solutions. The phone calls were marketed as “Free” nation-wide long distance calls. By the end of 1998 VoIP calls had yet to total 1% of all voice calls. By 2000, VoIP calls accounted for 3% and by 2003 that number had jumped up to 25% [2]. Broadband phone service was in full effect by 2005 as more VoIP providers emerged onto the scene. By this time, the issue of voice quality had been addressed with the increasing availability of high-speed Internet access. According to iDATE [2], by end of 2014, it forecasts 75 million mobile VoIP users in the US, UK, Germany and France.

In next, a brief introduction to the main elements relevant to VoIP system will be reviewed. VoIP protocols (RTP/RTCP and SIP) will be summarily presented, before proceeding to codecs.

2.2.1. VoIP system

In this section we review the elements that compose a VoIP system. Figure 2.2 reports the different block diagram involved in a communication under this system.

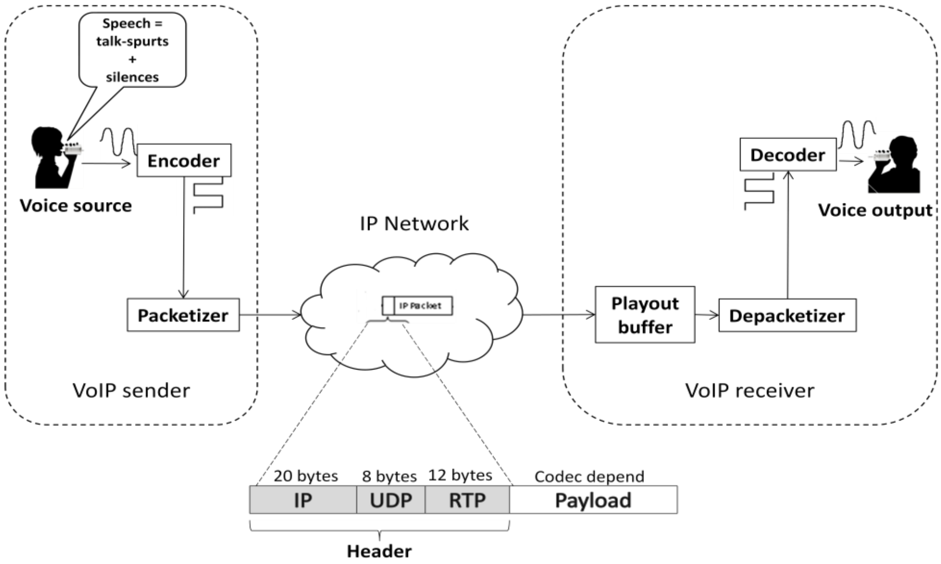


Figure 2.2: VoIP system architecture and VoIP Packet

- **Speech**: The speech source alternates between talking and silence period, which is typically considered to be exponentially distributed. A detailed explanation of speech models can be found in [1]. At this stage, we need devices to capture voice and transform it from a pressure wave to a continuous electric signal which can be processed by a computer. Conversely, the inverse procedure must take place at the reception point.
- **Digitalization**: Analog voice data is converted into a train of discrete samples and compressed using a *coder/decoder* (codec). We describe in detail voice codec in section 3.2.4.
- **Packetization**: The stream of binary data is, then, sent to the UDP/IP stack where it is broken into a series of packets of equal size preparing them for transmission across the network. Voice over IP is carried over RTP/UDP/IP packets. Figure 2.2 shows how the VoIP packet format is divided into the payload and headers. The headers associated with the VoIP packet are the *Internet Protocol* (IP) header, the *User Datagram Protocol* (UDP) header, and the *Real-time Transport Protocol* (RTP) header [3]. The voice payload size varies depending on codec used. When packet reaches the destination, it stripped of its headers and the payload is sent as a constant bit stream to a compatible codec (reverse process is performed).
- **Buffering**: Incoming VoIP packets will be buffered at the receiver. This later artificially delays their playout in order to compensate for variable network delays called jitter. VoIP packets are comprised by the playout buffer for certain duration depending on the codec deployed [4].

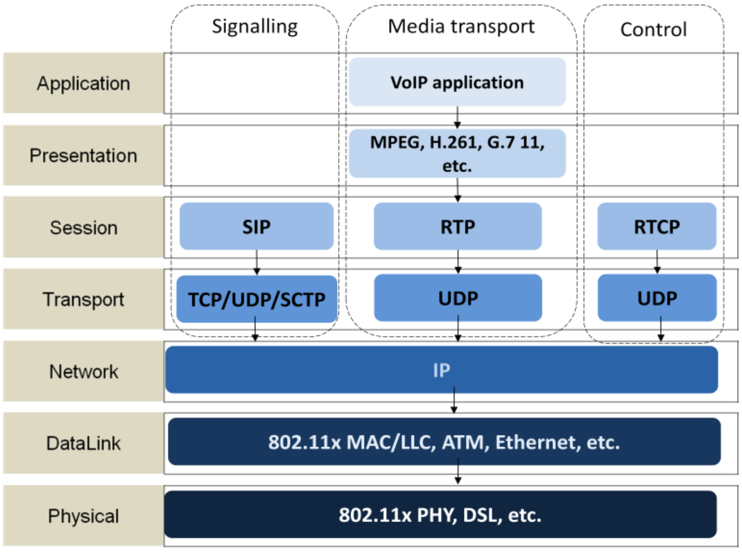


Figure 2.3: The VoIP protocol stack

2.2.2. VoIP protocols

It is important to note that VoIP works with any protocol stack that supports IP. According to VoIP protocol stack view (Figure 2.3), VoIP call consists mainly of two parts: signalling, media transmission and even controlling part.

- **Signalling**

this procedure allows call information to be carried across network boundaries by establishing the call, authenticating users, setting up the route, controlling the status of the call and terminating the session when the call is finished, negotiating which codec will be used, when the interchange of voice will start and end, to which port number the samples should be addressed, etc. The most used signalling protocols are the Session Initiation Protocol (SIP) [5] and the H.323 standard [10].

Session Initiation Protocol (SIP) is the IETF's standard (defined in RFC 2543) for multimedia session over IP. It has been increasing in success and acceptance among the VoIP community. SIP provides a simple and lightweight mechanisms for creating and ending connections for real-time interactive communications over IP networks, mainly for voice, but also for video conferencing, chat, gaming or even application sharing. However, It does not implement or control QoS, mobility management, or in general, any kind of application-specific media processing.

SIP User Agents use signalling message structures for call management. In the example of RFC 3261, there are two intermediary proxies. The SIP signalling messages for call setup are exchanged between two SIP phones via the two intermediary proxies. The call setup signalling messages are: INVITE, OK, ACK, BYE. The exchange of these four messages constitutes one SIP transaction. A transaction is a short sequence of SIP message exchanges. The successful calls show the initial signalling, the establishment of the media session, then finally the termination of the call.

- **Media transmission**

A media transport protocol has to be used in order to transport the encoded voice samples over a data network. Since VoIP is a real-time application, the chosen protocol has to show good properties in the face of delay, jitter and loss.

Real-time Transport Protocol (RTP) [6] is the media transport protocol of choice for most multimedia applications in the Internet. Originally developed by the Audio-Video Transport Working Group of the IETF [7], and specified in RFC 1889 (and updated in RFC 3550). RTP operates on top of UDP and defines a standardized packet format for delivering audio and video over the Internet. It brings the solution by complementing UDP's weaknesses. Basically, header includes information necessary for the destination application to reconstruct the original voice sample.

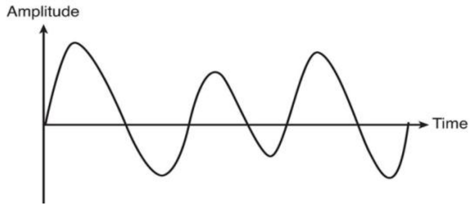
- **Controlling**

The control protocol companion continuously delivers quality feedback of the monitored session, provides an indirect way of detecting the effect that a PHY layer rate change has on it. RTP was designed together with its companion protocol, the Real-Time Control Protocol (RTCP). The role of this second protocol was to provide feedback on communication quality to the users, so that they could react accordingly. RTCP, on the other hand, provides extra information at the cost of more complexity. But since RTCP does not transport user data, it can accept non-real-time delay and processing times.

2.2.3. Voice codecs

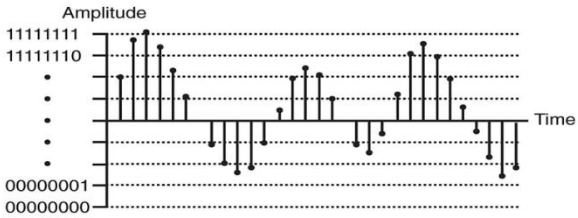
A *codec* (*co*der/*dec*oder) is a device and/or software program that is used typically to digitally encode an analog voice waveform (Figure 2.4 (a)). Coding process involves converting the incoming analog voice pattern into a digital stream and converting that digital stream back to an analog voice pattern at the ultimate destination. The objective of a codec is to obtain the lowest bit rate stream possible after conversion without degrading the quality of the signal so that the received audio signal can be generated without noticeable differences in quality [3].

Speech coding is a complex process, in which several degrees of liberty exist. As a consequence, a number of different algorithms have been devised [8]. Generally, these algorithms have to perform the same rough three steps:



(a) Analog voice waveform

(b) Waveform signal sampling



(c) Uniformly quantized signal

Figure 2.4: Voice waveform coding process

- **Sampling**: as shown in Figure 2.4 (b), Sampling is the process of converting the continuous analog waveform into a train of equally spaced, discrete samples which is taken at a fixed time interval. The sampling rate must conform to Nyquist's Theorem, which states that in order to be able to reconstruct the original analog signal, it must be sampled at a frequency equal to twice its bandwidth (typical sampling rate is 8000 Hz).
- **Quantization:** is the process of converting the height of the obtained samples to a finite number of discrete values (Figure 2.4 (c)). There are several methods to quantify classified according to their complexity (example: Uniform Quantization, Not uniform quantization). The difference between the actual signal and the digital reproduction is known as Quantization Noise.
- **Codification**: is the process by means of which a quantified sample is represented by a binary number (Figure 2.4 (c)). Not necessarily the sample value itself must be transmitted; other more sophisticated schemes can be used: For example, the difference between two samples could be coded, potentially reducing the number of bits needed. Or even the value of a sample could be used

by the receiver to predict the next one, eliminating the need to send it altogether.

There are several methods to digitize the voice samples. These methods vary by the information that is transmitted, the complexity of the algorithm, and the assumptions of the sound being transmitted (e.g. voice, fax, and music). In general, three codec families exist [9]:

- **Waveform codecs:** These codecs simply sample, quantize and send the information, without further considerations. Waveform coders are high-bit-rate coders (typically above 16 kbps). They are simple and provide very good quality, since they closely reproduce the original analog signal. Being so simple, they take low processing effort and hence do not introduce any additional delay into the system, which is an optimal characteristic for real-time communications. In exchange of this, they need fairly large bandwidth to provide good quality, and degrade rapidly otherwise. A well known example of waveform algorithms: Pulse Code Modulation (PCM), Differential PCM, and Adaptive Differential PCM.
- **Source codecs:** These codecs are called *vocoders*, operate using a mathematical model of the speech generation process at the human voice tract. Furthermore, since the generation of human voice presents a fair amount of correlation among consecutive samples, it is possible to predict the next samples from previous ones, with a high degree of probability. Hence, combining sample prediction at the receiver with the sending of only the filter parameters, the overall bandwidth needed can be much reduced. However, these sophisticated algorithms need much more processing effort than waveform codecs and generally use several samples at once in order to operate and predict the next ones, so that the overall effect is introducing some additional delay, as well as necessitating more powerful (and hence more expensive) signal processors. As an example of vocoders we find: Linear Predictive Coders (LPC).
- **Hybrid codecs:** are mixture of both previous techniques. These codecs use a mathematical model of the voice tract, but use a number of different input vectors to compare the result with the original signal. This way, a more precise encoding can be found for every sample. In this case, not only the filter parameters are sent but also an indication of which of the standardized excitation vectors has been used in generating it. As could be expected, these codecs lie somewhat in between the previous two in terms of bandwidth usage and quality. As an example of these algorithms: Code Excited Linear Prediction (CELP), Algebraic CELP, Low-Delay CELP, and Conjugate Structure Algebraic CELP.

Codec	Standard	Coding technique	Bitrates (Kbps)	Cost
G.711 [11]	ITU-T	PCM	64	Free
G.723.1 [12]	-	MPMLQ	6.3	Non-free
	-	ACELP	5.3	Non-free
G.726 [13]	-	ADPCM	16, 24, 32, 40	Free
G.728 [14]	-	LD-CELP	16	Non-free
G.729A [15]	-	CS-ACELP	8	Non-free
GSM-FR	ETSI	RPE-LTP	13	Free
GSM- EFR	ETSI	ACELP	12.2	Non-free
GSM- AMR	3GPP	ACELP	4.75 to 12.2	Non-free
iLBC	GIPS	LBC	15.2 , 13.33	Free
Speex	Xiph.Org	CELP	2 to 44	Free
Opus	IETF	CELT	6 to 510	Free

Table 2.1: Voice coding standards characteristics

The ITU-T (International Telecommunication Union - Telecommunication Standardization Sector) [16] was the first standards body to conduct early work on speech coding, and is the most active of the groups in this area. Table 2.1 lists a summary of several voice coding algorithms. As can be seen, there is a range of data rates available. The bit rate depends on the codec used and is the number of bits per second required to deliver a voice call.

Apart from reducing the header size, the size of the payload can also be reduced inside the frame. *Voice Activity Detection* (VAD) or Silence Suppression (SS) is a technique used in many codecs in order to reduce VoIP bandwidth. VoIP frames will not be generated continuously for each user, since there are silent periods and talk periods which follow an exponential distribution.

The QoS on VoIP network partly depends on the types of voice codec used [20]. VoIP applications require a strong QoS for the satisfaction of their users. last questions still arise: How must QoS for VoIP traffic can be defined? Can the user's perception of "good" or "bad" quality be accurately mapped to technical parameters like delay, jitter or packet loss? This was proven an elusive goal, and the next section reviews the most common indicators used to bridge that gap.

2.2.4. Speech quality assessment

Generally, the End-to-end quality of service control aims to increase the user satisfaction. With this goal in mind, the VoIP has been a heavily researched topic in QoS networking for more than one decade.

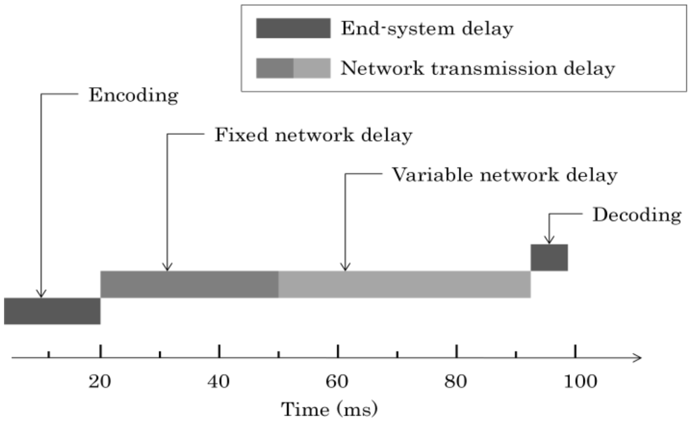


Figure 2.5: Components of one-way delay [21]

Normally, the speech signal is processed through the internet protocols and this one work on best effort based policy and leads to delay, jitter and packet loss during the communications. However, VoIP traffic is sensitive to these QoS parameters and needs some of the QoS requirements and recommendation to ensure acceptable voice quality.

2.2.4.1. QoS requirements

To understand voice and its requirements, we must understand its key performance indicators for QoS which are: delay, loss, and jitter.

- **Delay**

The delay is the key quality impairment for voice. End-to-end delay is the time spent from the moment when a frame is generated at the source until it is played at the destination. The following represents a set of delay recommendations from the ITU-T G.114 [17] specifications for one-way transmission time:

- 0 to 150*ms:* acceptable for most user applications.
- 150*ms* to 400*ms:* acceptable under awareness of impact on quality.
- Above 400*ms*: is unacceptable

For highly interactive tasks, quality may suffer at delays on the order of 100*ms*. In our experiment the maximum delay threshold will be 150*ms*.

The example time line in Figure 2.5 shows the component of delay, which impact the mouth-to-ear delay of VoIP service, using typical value for each component. In practice VoIP end system may incur an additional 5-20*ms* coded delay, depending upon the specific implementation (i.e. the codec delays are dependent upon the type of codec used) [21].

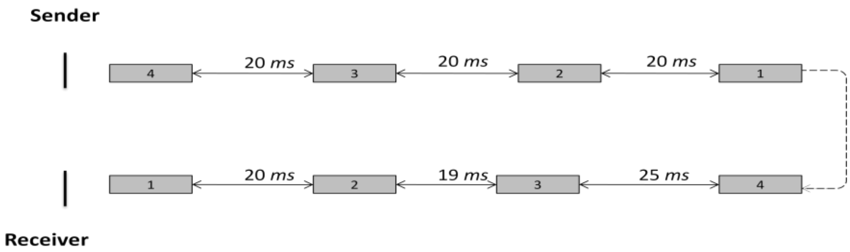


Figure 2.6: Delay variation at destination

- **Packet loss**

The packet loss is the percentage of the lost packet during the transportation. In general, due to network impairments, the voice packets are lost during communication and degrade voice signal at the receiver side. Additionally, much traffic in the network causes the network to drop packets. Yet, voice can afford a small amount of packet discard. A loss of 5% or more is usually noticeable. Though VoIP applications tolerate packet loss up to 10%, a packet loss of 1% still affects the quality of the VoIP stream [18]. In our experiment the maximum loss ratio will be 10%.

However, voice codec can handle loss in one of two ways. Interpolation is where the decoder uses sequence numbers in the packets to estimate and compensate for lost packets. The other method involves the encoder adding redundancy in the sent packets allowing reconstruction of lost packets.

- **Jitter**

Jitter or delay variation, is the difference between the minimum delay and the maximum delay that packets encounter in a single session. Figure 2.6 shows what the stream might look like at the receiving end where packets are generated each 20*ms* at the source side. In fact that the VoIP packets do not arrive precisely each 20*ms* that means we cannot play them out as they arrive unless we are willing to accept poor quality of the audio output.

Delay variations less than 75*ms* give good quality [21]. To overcome the problem of delay variation, buffering is used; once the destination starts receiving packets, it buffers them for a time period equal to the delay variation, and then starts playing them out. As a result, the end user receives smoothly played voice. The artificial delay until the play out can be either constant or variable over the lifetime of a call.

Several QoS evaluation models have been proposed with the goal of estimating/predicting the quality level of multimedia services according to the user's perception. Next, we present different approaches exist for evaluating the quality of voice packet.

2.2.4.2. Voice quality evaluation models

The first approach which is mainly of interest to service providers, involves using network performance metrics (measure of packet latency, loss, and delay variation, etc.). The second approach is objective and subjective quality measures which are of importance from the consumer's perspective to deduce his satisfaction.

Call quality testing has traditionally been subjective. In subjective testing methods, humans are asked to evaluate the quality of the service according to a standardized process and give a score, typically from 1 to 5 (MOS Score). While very accurate, subjective measures are very hard to conduct due to the constraints on the environment within which the tests take place and high number of subjects needed. That is why in the last few years more efforts have been focused on objective methods like the ITU-T E-Model. In the following we briefly describe both methods.

- **Mean Opinion Score (MOS)**

MOS is the original measure for speech quality which has been used in telephony networks to obtain the human user's view of the quality of the voice. Defined in ITU-T P800 specification [18], MOS was a subjective measurement where listeners would sit in a "quiet room" and score call quality as they perceive it. A listener is required to give each sentence a rating using a rating scheme which is the MOS score. From Table 2.2, MOS can range from 5 (Excellent) down to 1 (Bad). A typical range for acceptable Voice over IP quality would be from 3.6 to 4.2. A MOS below 3.6 results in many users who are not satisfied with the call quality.

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	annoying
1	Bad	Very annoying

Table 2.2: Mean Opinion Score (MOS)

Although it may be the best known voice quality tool, MOS is difficult to implement since human intervention is necessary (although estimates of voice quality can be made by automatic test systems). That is why ITU-T proposed a few years ago the E-model.

- **E-Model**

The E-model [19] provides a powerful and repeatable way for estimating the overall quality of VoIP calls. An E-model calculation considers all of the following VoIP factors: delay, packets lost ratio, delay introduced by the jitter buffer, and the behaviour of the codec. The output of an E-model calculation is a single scalar, called R-factor (Transmission Rating Factor). Once the R-factor is obtained, it can be mapped to an estimated MOS.

How to calculate R-factor?

The E-model is based on a mathematical algorithm. Combining a number of different impairments, The R-factor can be obtained through the following expression:

$$R = R_0 - I_s - I_d - I_e + A$$

Where: R_o represents the basic signal-to-noise ratio (SNR), I_s represents the simultaneous impairment factor, I_d represents the impairments caused by delay, I_e represents impairments caused by low bit rate codecs, is the equipment impairment factor representing impairments from low-bit codecs, (the so-called "equipment impairment factor") and packet loss, and A: is the advantage factor that models the user expectation of the technology employed. The typical range for the A factor is (0 – 20).

By assuming that impairments are mainly due to network condition, our interest is on the impairments due to codec, delay, losses and jitter introduced by the MANET, for that reason we can consider the simplified formula below given by Cole and Rosenbluth [22] to calculate the R-factor.

$$R = 94.4 - I_d(Ta) - I_e(loss, codec) - I_c(codec)$$

Where: I_d is a function of the absolute one-way delay (Ta), I_c is a function of the used codec type and packet losses, and I_e is the codec impairment (codec dependant).

With:

$$I_d = 25\{(1 - Ta^6)^{1/6} - 3(1 + \left[{}^{Ta}/_{3}\right]^6)^{1/6} + 2\}$$

Where:

$$Ta = log_2 \frac{delay\ (ms)}{100\ ms} \qquad \text{And} \qquad I_e = I_c + (95 - I_c). \frac{P_{pl}}{P_{pl}/(R_{Brust} + B_{pl})}$$

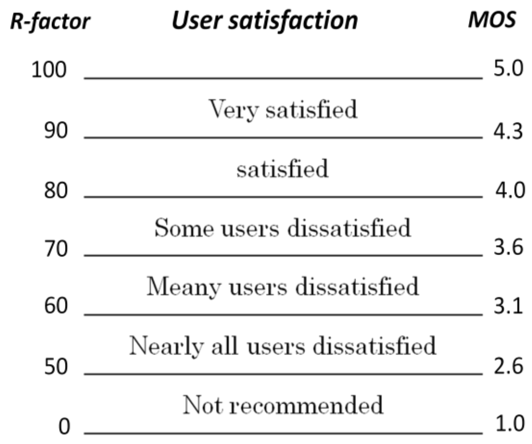


Figure 2.7: Relationship between R-factor and MOS values

Where: P_{pl}: Packet loss probability, *Bpl:* Packet loss robustness (codec dependant), *RBrust:* Models the burstiness of the losses

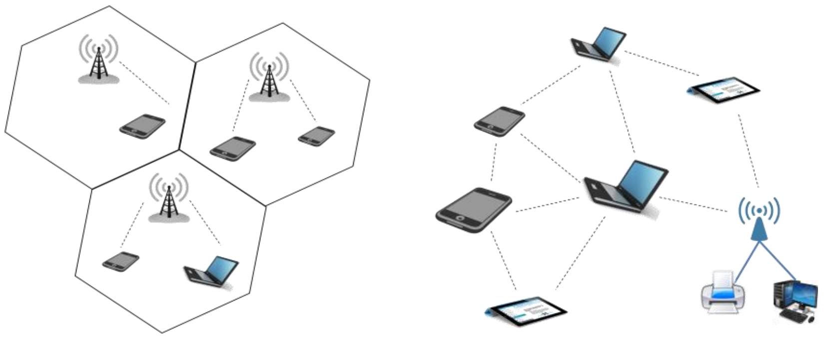
How to map between MOS and E-Model?

R factor values range from 100 (desirable) down to 0 (unacceptable) [19]. Once the value of R is calculated from these factors, an estimate of the MOS can be directly calculated using the following formula:

$$MOS = \begin{cases} 1 & R < 0 \\ 1 + 0.0035 + 7.10^{-6} R(R-60)(100-R) & 0 < R < 100 \\ 4.5 & R > 100 \end{cases}$$

The E-Model estimates how the average user would likely rate a call on an equivalent MOS scale (see Figure 2.7). R-Factor mapping to MOS method has the advantage of permitting a real-time calculation of the actual instant quality of service perceived by the user, as opposed to a fixed MOS value depending on the codec used and not taking into account factors that may vary during the call. Hence, the E-model will be used for obtaining the MOS in the simulation experiments presented later in this thesis, using ns2mesure tools [105].

Real-time voice transmission over mobile ad hoc network is very much demanding and necessary. In order to understand and assess VoIP quality under such dynamic networks, the Next section provides background material relevant to MANET and issues surrounding it constraints.



(a) Cellular network (b) Mobile ad hoc network

Figure 2.8: Cellular network versus mobile ad hoc network

2.3. Mobile ad hoc networks: Overview

2.3.1. General

In general, there exist three types of mobile wireless networks described as follows:

The infrastructure mode can be seen in Figure 2.8 (a), this mode comprises of wireless mobile nodes and one or more connecting centralize controller (base stations). A mobile node within the network looks for the nearest base station, connects to it and communicates with it.

Ad hoc mode does not have any infrastructure. It is devoid of base stations, routers and centralized administration, as depicted in Figure 2.8 (b). Nodes may move randomly and connect dynamically to one another. Thus, all nodes act as routers and must be capable of discovering and maintaining routes to every other node in the network and to forward packets accordingly. The last one is hybrid networks which combine infrastructured and ad-hoc aspects.

Mobile Ad hoc Networks (MANETs) are self-organising wireless networks with no fixed infrastructure or centralized control. As illustrated in Figure 2.9 MANETs consist of mobile nodes connected by wireless links and achieve communication by constructing a route over multiple wireless hops. This is why ad hoc networks are also referred to as multi-hop wireless networks. The network may operate in an isolated manner, or be connected to a fixed network (such as the Internet) through gateways. The logical nodes of such networks can be various objects, from very large objects (such as air planes, ships, trucks and cars), to very small objects (such as sensors). Correspondingly, the capacity and resource availability of the nodes vary significantly.

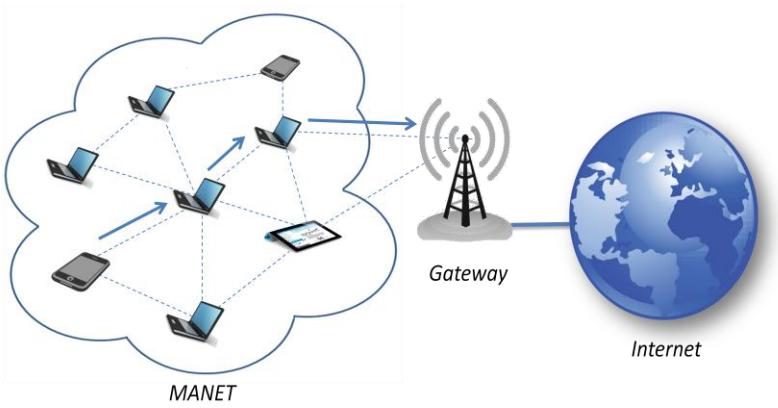


Figure 2. 9: Use scenario of a MANET

Node communication pattern could be described into two roles [23]:

- Each node is able to communicate directly with any node within its radio coverage area (single hop).
- Each node also implements the routing functionality in order to route (hop by hop) packets of an intermediate traffic. Especially, when two stations are not in the same neighbourhood, hence they cannot communicate directly. A multi-hop communication is initialized using intermediate nodes: Packets are forwarded from node to other before reaching the desired destination.

Despite the many design constraints, ad hoc networks offer numerous advantages that do not exist in cellular networks. Due to the lack of infrastructure and the unconstrained connectivity, they can be setup on demand, and can handle the rapid and unpredicted topology changes. Moreover, they are fault tolerant networks handling any malfunctions due to nodes movements, while maintaining the network operational. This takes place through network re-configurations carried out by efficient routing protocols. In the next Subsection, we provide a little history and present applications running on multi-hop ad hoc networks.

2.3.1.1. History background

Historically, ad hoc networks were initially introduced to improve communications in the military field. Civilian applications have appeared much later, in the late 1990s. The whole life-cycle of ad hoc networks could be categorized into the first, second, and the third generation ad hoc networks systems, which are the present ad hoc networks systems [23].

First generation: The concept of mobile ad hoc networking is not a new one and its origins can be traced back to the DARPA (Packet Radio Network project) in 1972. At the time, they were called PRNET (Packet Radio Networks). PRNET were used on a trial basis to provide different networking capabilities in a combat environment.

Second generation: in 1980s, ad hoc network systems were further enhanced and implemented as a part of the SURAN (Survivable Adaptive Radio Networks) program. This provided a packet-switched network to the mobile battlefield in an environment without infrastructure.

Third generation: For a long time, ad hoc network research stayed in the realm of the military, and only in the middle of 1990, the concept of commercial ad hoc networks arrived with notebook computers and other viable communications equipment. The IEEE 802.11 subcommittee had adopted the term "ad hoc networks" and the research community had started to look into the possibility of deploying ad hoc networks in other areas of application. Within the IETF (Internet Engineering Task Force), the Mobile ad hoc Networking working group (MANET) [24] was formed to standardize routing protocols for ad hoc networks. The development of routing within the working group and the larger community resulted in the invention of reactive and proactive routing protocols.

Currently, mobile ad hoc network research is a very vibrant and active field. The efforts of the research community, focus on standardizing different existing schemes for different network controls in a single framework which could be taken as a standard for all the future applications utilizing ad hoc networks as a networking technology.

2.3.1.2. Usages

The steadily wider adoption of wireless technologies in daily life let the integration of mobile ad hoc network applications. Hence, MANETs find its use in many civilian activities. Table 2.3 provides an overview of present and future common MANET applications, partially based on Reference [25].

2.3.2. Characteristics and complexities

Compared to other networks, ad hoc networks have the following special features that need to be addressed [23]:

Autonomous and Infrastructure-less: Network is self-organizing and is independent of any fixed infrastructure or centralized control. Therefore, the operation mode of each node is distributed peer-to-peer capable of acting as an independent router as well as generating independent data. This property yields some other special properties of ad hoc networks such as frequent topology change and limited resources.

Application	**Scenarios/Services**
Coverage extension	Extending cellular network access Linking up with the Internet, intranets, etc.
Emergency services	Search and rescue operations Disaster recovery Replacement of fixed infrastructure in case of environmental disasters Policing and fire fighting Supporting doctors and nurses in hospitals
Commercial and civilian Environments	Business: dynamic database access, mobile offices Sports stadiums, trade fairs, shopping malls Networks of visitors at airports
Vehicular services	Road or accident guidance, Transmission of road and weather conditions, Taxi cab network, Inter-vehicle networks
Sensor networks	Home applications: smart sensors and actuators embedded in consumer electronics Body area networks (BAN) Data tracking of environmental conditions, animal movements, chemical/biological detection
Home and enterprise networking	Home/office wireless networking Conferences, meeting rooms Personal area networks (PAN), Networks at construction sites
Education	Universities and campus settings Virtual classrooms Ad hoc communications during meetings or lectures
Entertainment	Multi-user games Wireless P2P networking Outdoor Internet access Robotic pets Theme parks Infotainment: touristic information

Table 2.3: MANET applications scenarios

Dynamic network topology: Due to arbitrary movement of nodes at varying speed, the network topology may change unpredictably, randomly and rapidly over time since nodes are free to move in an arbitrary manner. Furthermore, networks may be partitioned and merged from time to time because of node's movement and environment change. Generally, the mobility affects routing and network performance since the network must re-learn node locations after movement.

Wireless communication: Nodes use wireless interface to communicate. Wireless communication implies lower capacity as well as limited bandwidth, high collision probability, and high bit error rate. Many issues, which emerge from this fact such as multipath, path-loss, attenuation, shadowing, noise and interference on the channel, make the radio channel a hostile medium which behaviour is difficult to predict.

Limited node resources: Node resources, which include energy, processing capacity, and memory, are relatively abundant in the wired networks, but may be limited in ad hoc networks and must be preserved. For instance, Energy conservation becomes the major design issue as nodes in the MANET rely on batteries or some other exhaustible source of energy, and usually cannot generate power.

Limited physical security: The wireless mobile ad hoc nature of MANETs brings new security challenges to the network design. As the wireless medium is vulnerable to eavesdropping and ad hoc network functionality is established through node cooperation, mobile ad hoc networks are intrinsically exposed to numerous security attacks.

Scalability: As the number of nodes in an ad hoc network increases, the complexity of routing and configuration management also increases.

The specific characteristics of MANETs impose many challenges to network protocol designs on all layers of the protocol stack. For instance: the physical layer must deal with rapid changes in link characteristics. The media access control (MAC) layer needs to allow fair channel access, minimize packet collisions and deal with hidden and exposed terminals. At the network layer, nodes need to cooperate to calculate paths. Additionally, these features and their associated challenges make QoS provisioning design a very difficult task in such environment. Therefore, all these characteristics should be considered during a QoS management design. In the following sections, we present some important aspects of ad hoc wireless networks, especially in MAC and network layers.

2.3.3. Ad hoc MAC

The wireless access allows each node to only be a transmitter or a receiver at a time. Communication among mobile nodes is limited within a certain transmission range, and nodes share the same frequency domain to communicate. So, within such a range, only one transmission channel is used, covering the entire bandwidth. MAC (Medium Access Control) protocols are needed to regulate communication between nodes through a shared medium. It corresponds to the Data Link Layer (layer 2) of the ISO Open System Interconnect (OSI) reference model.

In mobile ad hoc networks, MAC protocols are needed as well to ensure successful operation of the network and play an important role in its performance. Thanks to the large availability in the market of inexpensive IEEE 802.11-based wireless devices, most of the research in ad hoc networks assumes the use of either the IEEE 802.11 MAC protocol. This wireless communication standard specifies two modes of MAC protocol: Point Coordination Function (PCF) mode, where mobile terminals communicate with (and through) one or more access points, and Distributed Coordination Function (DCF) mode, where mobile nodes are allowed to communicate and to interact directly, without using any infrastructure. In DCF protocol, all the stations compete with each other to gain access to the channel. This protocol is based on CSMA/CA which is a contention-based protocol, making certain that all stations first sense the medium before transmitting. The main goal is to avoid having stations transmit at the same time, which results in collisions and corresponding retransmissions. To avoid collisions, the station uses two types of sensing: physical carrier sensing and virtual carrier sensing.

As an optional feature of 802.11 is the RTS/CTS (Request to Send/Clear to Send) function to control station access to the medium. The primary reason for implementing RTS/CTS is to minimize collisions among hidden stations. If the channel is free, node will transmit a request to send (RTS) to the destination. The destination will respond with a clear to send (CTS) if it is available to receive data (i.e. if it is not receiving data from another node). When the source node receives the CTS, it will transmit its data. Along with both the RTS and CTS, a network allocation vector (NAV) is transmitted, which is explained below. After correct reception of the data, the destination will transmit an acknowledgment (ACK) back to the sender. At this point, if the sender has more data to transmit, it will again begin its *backoff* and repeat the process. This process is demonstrated in Figure 2.10.

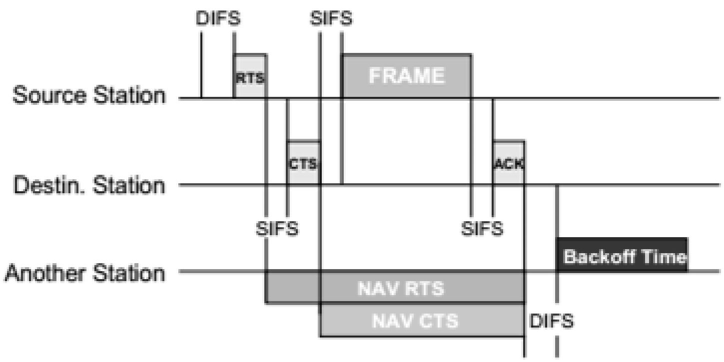


Figure 2.10: Virtual Carrier Sensing mechanism

2.3.4. Ad hoc routing

Routing is the process of choosing a path in a network for moving packets form source to destination. It basically involves two processes: finding an optimal routing path and transfering the packets in the internetwork. However, the potential problem to this mechanism is some destinations might be unreachable. Hence, a routing protocol is needed which is the software or hardware implementation of a routing algorithm. This protocol uses deferent metrics to select a path to transmit a packet across an internetwork. These metrics include: hop count, Bandwidth, Delay, Load, Cost (in terms of Energy Consumption and Time process), etc.

Routing in a mobile ad hoc network is different than routing in an infrastructure-based network, because MANETs are characterized by a multi-hop network topology that can change frequently due to mobility. Efficient routing protocols are needed to establish communication paths between nodes, without causing excessive control traffic overhead or computational burden on the power constrained devices.

One of the most popular methods to distinguish mobile ad hoc network routing protocols is based on how routing information is acquired and maintained by mobile nodes. A large number of solutions have already been proposed, some of them being subject to standardisation within the IETF. Using these solutions, mobile ad hoc network routing protocols can be divided into proactive routing, reactive routing, and hybrid routing [25].

Proactive routing protocols: also called a "table-driven" routing protocol, are typically modified versions of traditional link state or distance vector routing protocols encountered in wired networks, adapted to the specific requirements of the dynamic mobile ad hoc network environment. Proactive approaches attempt to have an up-to-date route to all nodes at all times. To this end, these protocols exchange routing control information periodically so will as on topological changes.

Proactive routing protocols may waste bandwidth since control messages are sent out unnecessarily when there is no data traffic. The main advantage of this category of protocols is that hosts can quickly obtain route information and quickly establish a session. Examples of proactive routing protocols for ad hoc networks include Destination Sequenced Distance Vector (DSDV) [26], Topology Based dissemination based on Reverse Path Forwarding (TBRPF) [27], and Optimised Link State Routing (OLSR) [28] which will be described in detail in Subsection 2.3.6.

Reactive routing protocols, also called "on-demand" routing protocols, only set up routes to nodes they communicate with, and these routes are kept alive as long as they are needed. Cause in most of the time, it is not necessary to have an up-to-date route to all other nodes. A route discovery operation invokes a route determination procedure. The discovery procedure terminates when either a route has been found or no route is available after examination for all route permutations. In a mobile ad hoc network, active routes may be disconnected due to node mobility. Therefore, route maintenance is an important operation of reactive routing protocols.

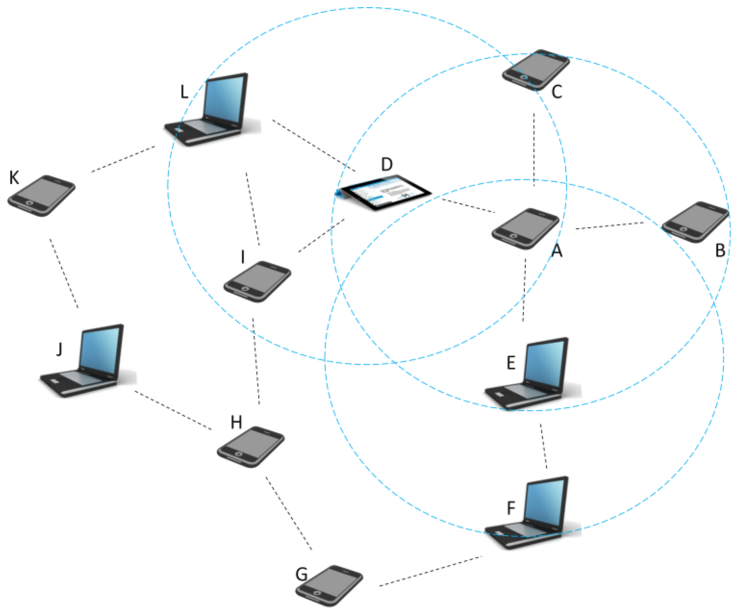
Reactive routing protocols can dramatically reduce routing overhead because they do not need to search for and maintain the routes on which there is no data traffic. This property is very appealing in the resource-limited environment. Some examples of this type of protocol are Dynamic Source Routing (DSR) [29] and Ad-hoc On Demand Routing (AODV) [30].

Hybrid routing protocols is a combination of proactive and reactive protocols, where nearby routes (for example, maximum two hops) are kept up-to-date proactively, while far-away routes are set up reactively. These protocols are designed to increase scalability by allowing nodes with close proximity to work together to form some sort of a backbone to reduce the route discovery overheads. An example of hybrid routing protocol is Zone Routing Protocol (ZRP) [31].

Other routing protocols approach has been considered, such as location-based routing protocol which is completely different. In this approach, packet forwarding is based on the location of a node's communication partner. Location information services provide nodes with the location of others, so packets can be forwarded in the direction of the destination. Examples of protocols using these techniques are: Greedy Perimeter Stateless Routing (GPSR) and Distance Routing Effect Algorithm for Mobility (DREAM).

Reference [32] provides an extensive overview of routing protocol research. Simulation studies have revealed that the performance of routing protocols in terms of throughput, packet loss, delay and control overhead strongly depends on the network conditions such as traffic load, mobility, density and the number of nodes.

Figure 2.11 provides an overview in terms of routing table content of the most popular unicast routing protocols. One advantage of proactive MANET routing protocols is that they tend to provide lower route discovery latency than on-demand protocols because they maintain route information to all the nodes in the network at all time. This feature makes proactive routing attractive protocols for real-time applications. The next section gives background information on signalling in proactive routing protocols. Especially, we survey some key concepts on generic soft-state signalling in MANET. Furthermore, we illustrate the detailed signalling mechanisms used in the OLSR routing protocols in terms of neighbour detection and topology advertisement



Routing Type	Routing knowledge of node A when communicate with nodes B,F and J	Protocol
Proactive	A has route to all nodes in the network	DSDV, OLSR, TBRPF
Reactive	A has only route to B, F and J	AODV, DSR
Hybrid	A has route to B, C, D, E, F, I and L proactively and route to J reactively	ZRP
Location based	A knows at least the location of its neighbours B, C, D, E, and the location of F and J	GPSR, DREAM

Figure 2.11: Overview in terms of routing table content of existing routing protocol

2.3.5. Signalling in proactive MANET routing protocols

Network Signalling is defined as the exchange of control information among elements of a telecommunication network to initiate, maintain, and release connections and for network management [33]. Signalling protocols define a standard set of information elements and a method of transport in order to enable components of a network to interoperate. These protocols can be classified as either soft state-based, hard state-based or a mixture of both.

In soft state-based approach, nodes maintain state consistency through periodic refresh messages. Simplicity and robustness are two features which make this mechanism an attractive choice for a variety of network protocols and applications, especially for dynamic networks such as MANETs. For instance, established links maybe get broken in MANET due to node mobility or failure. If nodes are unaware of such changes and keep forwarding data packets towards these broken links, the data packets could be lost. Hard state-based approaches use reliable messages to install a state and keep the state unless it is explicitly removed by a teardown message, while a soft state can expire after a certain period of time if not refreshed.

Proactive signalling exchanges signalling messages periodically, based on soft-state mechanisms. It just inserts new state into signalling messages then the receivers of such messages check each state and update message against their state repositories. Proactive Soft state Signalling system is based on two principal functions:

- *Neighbour Detection*: In most MANET routing protocols, either proactive, reactive or even hybrid protocols, periodic HELLO messages are exchanged between neighbouring nodes to detect link dynamics and to maintain node status. For example, nodes running OLSR discover new neighbours and links when receiving the first HELLO message from an *unknown* neighbour. Moreover, obsolete neighbour states (caused by for example link breakage) are removed after state time-out.

- *Topology Advertisements:* Proactive routing protocols like OLSR, TORA and TBRPF propagate periodic network-wide *topology update* messages to advertise topology changes. In addition to initiating new link state in topology repositories of each node, for example, the topology advertisement process in OLSR removes obsolete topology state either implicitly by assigning sequence numbers to topology advertisements, or by state time-out.

Timer Intervals: Soft state protocols have two timers associated with the control traffic, one at the sender and one at the receiver. Refresh timer maintained at the sender side used to clock out the refresh messages for the existing state. Timeout timer maintained at receiver side which discards a state entry if it does not receive a refresh message for that state before the timeout timer expires

One of the concerns of the soft state approach is configuring Timer Intervals. Traditional protocols have fixed settings for the timer values. The value of the refresh and timeout timers is mainly determined based on recommendations of original protocol designers or empirical observations. Consequently, the value of such intervals usually remains fixed no matter what network conditions (i.e. node velocity, link loss rate, QoS provisioning) are. Therefore, several questions as to the configuration of timer intervals may arise: e.g. in voice transmission over MANET context. Does the default value of timer intervals work well against all types of audio codecs?

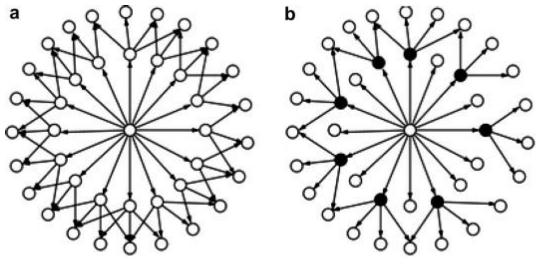


Figure 2.12: Comparison of full flooding (a) and MPR flooding (b)

2.3.6. Proactive soft-state signalling in OLSR

2.3.6.1. Overview

OLSR, presented in 1998 [28], concentrates on routing in ad hoc networks. It is a table-driven proactive routing protocol for MANETs based on the link state approach and uses the Shortest Path First (SPF) algorithm. Due to its proactive nature, it has the advantage of making the routes ready before they are needed by exchanging topology information with other nodes of the network periodically. OLSR provides the following optimizations to the classic link state algorithm:

- It reduces the size of control packets by implementing Multipoint Relays (MPRs).
- It minimizes flooding of control traffic by only permitting select nodes, MPRs, to send control traffic through the network, and does not generate extra control traffic in response to link failures or arriving nodes.

Each node has to maintain a list of nodes qualified as MPRs. MPRs are used in the flooding mechanism in order to reduce the flooding process and to periodically maintain the topology. Figure 2.3 compares MPR flooding to full flooding as used by classic flooding mechanisms.

2.3.6.2. State Maintenance

OLSR simply relies on the soft-state approach to maintain the consistency of topology information among nodes in the network. Apart from the periodic control messages, OLSR does not generate extra control traffic in response to link failure and nodes joining/leaving events. OLSR deploy tow essential signalling messages: HELLO Messages and TC messages.

- *Hello messages:* are sent only for one hop and serve to discover neighbours, which are localized in the range of the local node. Links can be symmetric or asymmetric; OLSR considers that two nodes which are two-hop neighbour (via another node) can not behave as neighbour even if links are symmetric.

The HELLO message contains a list of neighbours and the status of each corresponding link. For this reason, HELLO messages are considered as an immediate informative packet which updates the quality of links of the neighbourhood of a given node. In addition, when nodes leave the network, or the links between them get broken, the corresponding link state and the neighbour state is removed after the state timeout timers expire without receiving any HELLO messages.

- *TC messages (Topology Control):* Each node periodically broadcasts TC messages to declare its MPR selector set and populate its topology table. Node records information about the topology of the network as obtained from TC messages. Based on such information, the routing table is calculated based on the SPF algorithm. In addition, periodic TC messages help the remote nodes recover from loss of topology information caused by state inconsistencies or node restarts.

A routing table is kept at each node and contains routes to all other destinations in the network. This table is built by tracking connected pairs (i.e., pairs which link status is bi-directional) in the topology table. In order to obtain optimal paths, only connected pairs are selected on the minimal path. There is no entry for destinations whose routes are broken or are not fully known. Route table entries contain the destination address, next-hop address, and estimated distance to destination (in number of hops).

2.3.6.3. OLSR Timers

In OLSR, the *soft state timers* have two types of usage: message generation and state maintenance. Message generation timers (HELLO and TC interval timers) are used to send periodic HELLO and TC messages, while state-maintenance timers are to keep updated the state information in OLSR internal tables and remove obsolete state by time-out.

Each OLSR node has four timers associated with the control traffic.

- *HELLO refresh timer* (HELLO INTERVAL) is used to clock out the HELLO messages to refresh existing state and advertise new neighbour state.
- *TC refresh timer* (TC INTERVAL) is the timeouts before resending TC messages used to refresh existing topology state and advertise topology changes.
- *Neighbour state timeout timer* (NEIGHB HOLD TIME) is to remove a neighbour state entry if the node does not receive a HELLO message for that state before the state timeout timer expires.

- *Topology state timeout timer* (TOP HOLD TIME) is used to remove a topology state entry if the node does not receive a TC refresh message for that state before the state timeout timer expires.

By default, the neighbour holding time (i.e. the interval of the neighbour state timeout timer) is set to be 3 times the value of the HELLO interval. The topology information holding time (i.e. the interval of the topology state timeout timer) is three times the default value of the TC interval. Table 2.4 shows OLSR timers with their standard values as specified in RFC 3626. Each parameter can take a range of values, this range has been defined here by following OLSR restrictions, with the aim of avoiding pointless configurations.

Parameter	**Standard configuration**	**Range**
HELLO_INTARVAL	0.2 s	1 ...30
TC_INTERVAL	5.0 s	1 ... 30
NEIGHB_HOLD_TIME	3 × HELLO_INTERVAL	3 ... 100
TOP_HOLD_TIME	3 × TC_INTERVAL	3 ... 100

Table 2.4: Default OLSR timers parameters values following the RFC 3626 specification

From a technological point of view, the realization of a mobile ad hoc network platform requires a large number of challenges to be solved related to devices, protocols, applications and services. Despite the large efforts of the MANET research community and the rapid progress made during the last years, a lot of challenging technical issues remain unanswered. Among these issues is the Quality of Service (QoS) provisioning. The next section overviews several approaches proposed for QoS in mobile ad hoc networks.

2.4. Quality of Service management in MANETs

According to ITU-T Recommendation E.800 [34], QoS has been defined as: "The collective effect of service performance which determines the degree of satisfaction of a user of the service". QoS is a measure of the quality a user can expect from a given service provided by a network. It refers to traffic control mechanisms that seek to either differentiate performance based on applications which all have their own specific demands to the QoS. There must be an interaction between what the applications demand from the network and the services that the network provides. Taxonomy of QoS specifications given in [35] is a widely cited source for discussing QoS parameters.

2.4.1. QoS metrics

Metrics are quantitative / qualitative indicators suitable to point out the ability of a system to accomplish its intended service according to its specification or user requirements. Different network applications will have their own QoS demands to specify QoS requirements. Consequently, QoS metrics may be used as constraints on different layers (i.e. network layer as route discovery and selection). We describe in the following the commonly used metrics for QoS demands from real-time applications.

Bandwidth/Throughput: Applications have different needs for bandwidth and require a certain amount of rate to be offered by the network. The bandwidth available in the network is finite, so the different applications compete for the available bandwidth resources. Throughput is generally measured as the percentage of successfully transmitted radio-link level frames per unit time. It is the actual rate offered by the network, and hopefully this satisfies the bandwidth required for the specific application.

End-to-end delay: is defined as the interval between the frame arrival time at the MAC layer of a transmitter and the time at which the transmitter realizes that the transmitted frame has been successfully received by the receiver. Another point of view is the amount of time it takes for a packet of data sent from a source port until it reaches its destination port. End-to-end delay consists of the following components:

- Packetization delay: refers to the time it takes at the source node to collect all bits that compose a frame for sending over the network.
- Queuing delay: represents the time packets spend in the intermediate nodes waiting to be forwarded.
- Transmission and propagation delay: is the time it takes to clock out a packet to a link and to propagate it over the link.
- Play-out delay: the time packets spend in the buffer of the destination for smooth play out.

Jitter: This metric can also be expressed as delay variance. Jitter is defined as the difference between the upper bound on end-to-end delay and the absolute minimum delay. The former incorporates the queuing delay at each node and the latter is determined by the propagation delay and the transmission time of a packet. Common jitter-sensitive applications are streaming video and voice.

Packet loss: The acceptable percentage of total packets sent, which are not received by the transport or higher layer agent at the packet's final destination node; while traversing a network, packets of data may pass through multiple processing stages as they are routed from source to destination. Bottlenecks may arise, resulting that packets are filling queues faster than they are being forwarded out. For the network to remain stable, the only solution is to drop packets (packet loss).

Other metrics can be employed by routing protocols for path evaluation and selection in order to improve all-round QoS or to meet the specific requirements of application data sessions in MANETs. Among them the NRL (**Normalized Routing Load**) which is defined as the ratio of the number of control packets propagated by every node in the network, to the number of data packets received by the destination nodes.

The following equations are analytical models for QoS metrics in MANET context. The MANET can be expressed as a weighted directed graph $G(V,E)$, where V is a set of nodes with a wireless connections E. If there are n+1 nodes V, $V = \{v_0, v_1, v_2, v_3, \ldots, v_n\}$. The communication radius of each node is r_i, its communication area is A_{vi}, and the edge $e = (v_i, v_j) \in E$ denotes the two-way wireless connection between two nodes (v_i, v_j). The QoS metrics of the path $P(v_s, v_d)$ can be calculated. Given the source node $v_s \in V$ and the destination node $v_d \in V$, the corresponding QoS metrics of path $P(v_s, v_d)$ are computed as the following:

$$Bandwidth(P(v_s, v_d)) = min_{e \in P(v_s,v_d)} \ \{Bandwidth(e)\} \qquad (1)$$

$$Delay\big(P(v_s, v_d)\big) = \textstyle\sum_{v \in P(v_s,v_d)} Delay(v) + \sum_{e \in P(v_s,v_d)} Delay(e) \qquad (2)$$

$$Packet_{loss}\big(P(v_s, v_d)\big) = 1 - \textstyle\prod_{e \in P(v_s,v_d)} (1 - Packet\_loss(e)) \qquad (3)$$

The QoS metrics could be concave or additive. Bandwidth is concave in the sense that end-to-end bandwidth is the minimum of all the links along the path (equation 1). End-to-end delay is additive because it represents the accumulation of all delays of the links along the path (equation 2).

With the evolution of wireless communications and the emergence of diversified multimedia technologies, an application may typically request a particular quality of service by specifying its requirements in terms of one or more of the above metrics. For example, it may require a guaranteed throughput of 500 *kb/s* and a maximum packet delay of 50 *ms*. In most cases, the QoS protocol should only admit this data session into the network if it can provide the requested service. Besides existing problems for QoS in IP networks, the characteristics of MANETs impose new constraints due to the dynamic behaviour of each host and the variation of limited available resources.

2.4.2. QoS provisioning approaches for MANETs

Quality of service management in ad hoc networks has become an area of great interest. However, QoS support affects most of the layers in the OSI-model. Physical layer, link layer, network layer, transport layer and application layer have all influence when supporting QoS across the protocol layers. QoS management approaches in mobile ad hoc networks can be classified depending on at which layer of TCP/IP protocol suite, that we call *layered classification*. Generally, we can classify methods into the following layered categories: MAC layer, Network layer and Cross layer.

2.4.2.1. MAC layer level: differentiated service

One challenge in supporting QoS for real-time applications is associated with the design of the medium access control (MAC) protocol. In order to support QoS in MANETs, some level of service differentiation at MAC level is needed to guarantee applications requirements. The IEEE 802.11 standard is the most widely used MAC layer for mobile ad hoc networks. There are two major flavours of IEEE 802.11 pertaining to coordination function for MANETs: the first is with Distributed Coordination Function (DCF) and the second is with Enhanced Distributed Channel Access (EDCA). The version with EDCA is also called IEEE 802.11e [36] and has a support for the provision of QoS.

IEEE 802.11 DCF is a good example of best-effort type control algorithm. It has no notion of service differentiation and no support for real-time traffic. Veres et al. [37] has proposed a scheme to extend IEEE 802.11 DCF with ability to support at least two service classes: High-priority (i.e. premium service) and best-effort. Traffic of premium service class is given lower values for CW_{min} and CW_{max} than those of best-effort traffic. Analytical results show that, by setting different values of *CW*, differentiated levels of service can be achieved. A comprehensive simulation is also conducted which shows similar results.

The upcoming 802.11e standard [36] enhances the current 802.11 Medium Access Control to support applications with QoS requirements. It provides a channel access function, called Hybrid Coordination Function (HCF). The HCF uses both contention-based channel access methods, called Enhanced Distributed Channel Access (EDCA) mechanism for contention based data transfer. The EDCA provides differentiated and distributed access to the wireless medium. This differentiation is achieved by varying the amount of time a node would sense the channel to be idle and the length of the contention window during a *backoff*. Each frame from the higher layer carries its user priority (*UP*). The EDCA supports eight different *UPs*. After receiving a frame, the MAC layer maps it into one of the four *ACs*.

The design of an efficient MAC for MANETs that support QoS always remains a major problem that needs to be addressed. Although, there are extensions of CSMA/CA and IEEE 802.11 for supporting QoS, however, a node that wishes to transmit may need to wait for a random amount of time due to the exponential *backoff* mechanism. Also, due to hidden and exposed terminal problems, neighbours in the carrier sensing range cannot transmit simultaneously. Moreover, due to intra-path correlation, neighbouring nodes that are lying along a path from the source to the destination cannot transmit simultaneously and thus the effective bandwidth of the path is less than the raw bandwidth of a link.

Most of the existing MAC schemes focus only on a subset of QoS features with a simple network topology, while ignoring the issues of end to end packet delay in multi-hop networks, channel errors, power control, heterogeneous nodes, node mobility, etc. Furthermore, they consider one flow per node where all the packets have same priority. In multi-hop ad hoc network, a node may be forwarding packets belonging to different flows, which may have very different bandwidth, delay bounds and priority. Works on differentiation service at MAC level led to interesting results but insufficient. It is indispensable to associate mechanisms operating at higher levels (network, transport and application) to ensure a high quality of service.

2.4.2.2. Network layer level: Routing protocol with QoS

Initially, most routing protocols for MANET, such as AODV, DSR, and OLSR, are designed without explicitly considering quality-of-service of the routes they generate. QoS routing in MANETs has been studied later [38]. QoS routing requires not only finding routes from a source to a destination, but routes that satisfy the end-to-end QoS requirement, often given in terms of bandwidth or delay. However, QoS is more difficult to guarantee in ad hoc networks than in most other type of networks, because the wireless bandwidth is shared among adjacent nodes and the network topology changes as the nodes move.

Classical routing schemes were proposed for routing messages on the shortest available path or within some system-level constraints. Routing messages in such paths may not be adequate for applications that require QoS support. The characteristics of good QoS routing algorithms include the following aspects [39]:

- The routing algorithm should compute an efficient route that can satisfy the QoS requirements with a very high probability, if such a route exists.
- The QoS route computation algorithm should be simple and robust.
- With the change in network dynamics, the propagation and updates in state information should be kept to a minimum.

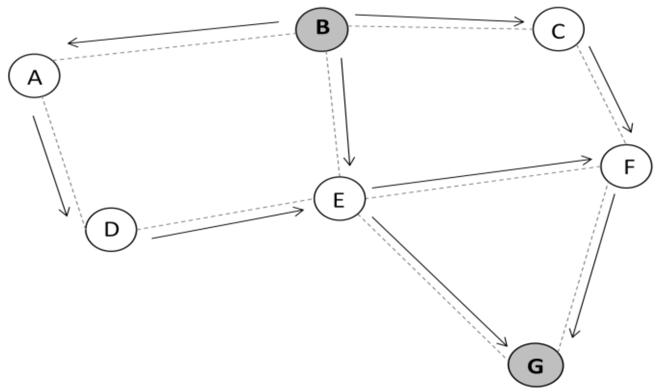


Figure 2.13: Available paths from node B to node G

No.	Path	Hop count	BW (Mbps)	Delay (ms)
1	$B \rightarrow E \rightarrow G$	2	2	9
2	$B \rightarrow E \rightarrow F \rightarrow G$	3	2	11
3	$B \rightarrow C \rightarrow F \rightarrow G$	3	4	15
4	$B \rightarrow C \rightarrow F \rightarrow E \rightarrow G$	4	3	19
5	$B \rightarrow A \rightarrow D \rightarrow E \rightarrow G$	4	2	23
6	$B \rightarrow A \rightarrow D \rightarrow E \rightarrow F \rightarrow G$	5	2	25

Table 2.5: Attributes of the available links between B and G

For example, Figure 2.13 shows a wireless topology example of QoS routing. The mobile nodes are labelled as $A, B, C, \ldots,$ and G. Suppose we want to find a route from a source node B to a destination node G. Here, 6 paths are available between nodes B and G. Table 2.5 shows the attributes of each link, in the table *BW* and *Delay* represent respectively available bandwidth in Mbps and delay in milliseconds of the wireless links.

For conventional routing using a shortest path (in terms of the number of hops) as metric, the route path “1” will be chosen. It is quite different in QoS route selection. Suppose a packet-flow from node B to node G requires a bandwidth guarantee of 4 Mbps. Now the feasible route will be the path “3”. The shortest path route “1” will not be suitable for providing the required QoS. Hence, QoS routing has to select a suitable path that meets the QoS constraints specified in the service request made by the user. The end-to-end delay of a path is equal to the sum of delays of all the links of a path. Clearly path “3” is not optimal in terms of hop count and/or end-to-end delay parameters, while path “1” is optimal in terms of both hop count and end-to-end delay parameters.

Several protocols are proposed in the literature in which the provision of QoS in an ad hoc network is embedded in the routing protocol itself. The provision of QoS at the level of routing can be divided into different categories for the sake of simplifying, some use generic QoS measures and are not tuned to a particular MAC layer . The QoS routing protocols proposed for MANETs can be classified under different criteria:

- Routing information update mechanism (Proactive/table-driven, Reactive/On-demand, Hybrid)
- Use of information for routing (Information of past history, Prediction)
- State maintenance (Local, Global)
- Routing topology (Flat, Hierarchical)
- Interaction with MAC layers (Independent, Dependent)
- Number of Path Discovered (Single path, Multiple paths)
- Utilization of Specific Resources (Power aware routing, Geographical information assisted routing)

A general survey on QoS routing in MANET is presented in [39], while some of the main issues that must be addressed by QoS routing solutions are suggested in [40]. The [39] describes some representative protocols with various unique features for providing QoS support at the routing level. Each of these protocols addresses the problems of bandwidth/delay estimation, route discovery, resource reservation, and route maintenance in a unique manner, providing various advantages and disadvantages for each protocol. In what follows, Table 2.5 classifies some of them.

Generally, in reactive protocols such as AODV, the routes are computed on demand. Thus, the applications can specify their constraints on the QoS metrics, and the routing protocol will search for a route that satisfies them during its route discovery process. Regarding proactive routing protocols, contrary to reactive ones the routes are computed proactively and prior to application requests. Thus, proactive protocols will need to select the route achieving the best value since it does not currently know the application requirements. In this work, we will focus on the proposals for quality of service using the proactive Optimized Link State Routing protocol.

Protocol	QoS metrics	Route discovery	MAC dependent
CEDAR [41]	Bandwidth	Hybrid	Yes
QOLSR [42]	Bandwidth, delay	Proactive	None
QoSAODV [43]	Bandwidth	Reactive	None

Table 2.6: Comparison of QoS routing protocols

QoS routing is key issue in provision of QoS in Ad Hoc networks. Number of QoS routing approaches have been proposed in literature, focusing on different QoS metrics, and no particular protocol provides overall solution. Routing protocols with QoS operating with MAC protocol uses probabilistic estimation of available resources. Typically, they estimate the state of nodes and links. However, they can only provide slight QoS guarantees since no promises can be made to applications.

2.4.2.3. Cross-layer QoS strategies

Cross-layering technique [44] is considered as one of the effective methods to enhance the performance of a wireless network by jointly designing multiple protocols. These methods address both cellular and ad hoc networks. Cross layer design uses the properties or functionalities of more than one layer by Involving different layers of the protocol stack, and allowing parameters to be passed to the adjacent layers to assist them in determining the operation modes that will suit some requirements imposed by the nodes.

A layered architecture, like the seven-layer open systems interconnect (OSI) model, divides the overall networking task into layers and defines a hierarchy of services to be provided by the individual layers. The services at the layers are realized by designing protocols for the different layers. The creation of new interfaces between the layers is not available in the layered architecture. The layered architecture forbids direct communication between nonadjacent layers; communication between adjacent layers is limited to procedure calls and responses. Therefore, cross-layer designs are used for information sharing between the layers at runtime. Different designs of cross-layer architectures have been proposed in the literature. These can be put into three categories as shown in Figure 2.14:

- Direct communication between layers (Figure 2.14 (a)): it allows layers to communicate with each other in order to straightforward way to share runtime information between them.
- A shared database across the layers (Figure 2.14 (b)): the common database (like a new layer) provides the service of storage/retrieval of information to all the layers. An optimization program can interface with the different layers once through the shared database.
- Completely new abstractions (Figure 2.14 (c)): authors in [45] present a new way to organise the protocols in heaps, not in stacks as done by layering.

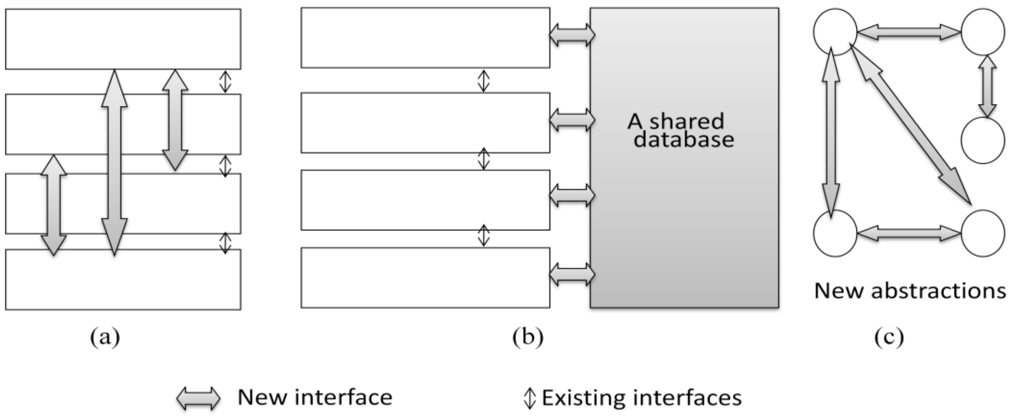


Figure 2.14: Cross-layer architectures design [44]

Cross-layer design is extremely important in supporting QoS in MANETs due to the shared media and distributed organization of the network. For example, the fact that the wireless channel is shared among neighbours makes the estimation of available resources extremely difficult. Moreover, in contrast to layered architecture technique, cross-layering allows communication between non-neighbouring layers as well as reading and controlling parameters of one layer from other layers [46]. This layers collaboration can help with the processes of resource estimation. Similarly, feedback from the network layer to the application on available resources provides applications the opportunity to adjust their transmission appropriately.

Several works have been considered the cross-layer design for MANET. This approaches aim to optimize overall network performance by increasing local interaction among protocols, decreasing remote communications, and consequently saving network bandwidth. In addition, the shared cross-layer information can be used to construct a local view of node state. This information can be used as knowledge for adaptation decisions if necessary. However, Works on cross-layer architecture approaches have not reached a maturity allowing them to be used as reference models.

MANET constraints pose major challenges to QoS provisioning. As a result, it is necessary to improve overall network performance to achieve a successful deployment of a MANET on a large scale and in different application areas. We have presented several approaches dealing with this issue at different levels (MAC, routing, cross-layer). However, the management of QoS in MANETs is still a key issue to be coped with. This problem becomes more difficult with applications requiring a high level of QoS, like real-time applications. In the next section, we switch to schemes that are part of the network management classifications. We discuss the research reported in the literature related to network management architectures designed for MANET.

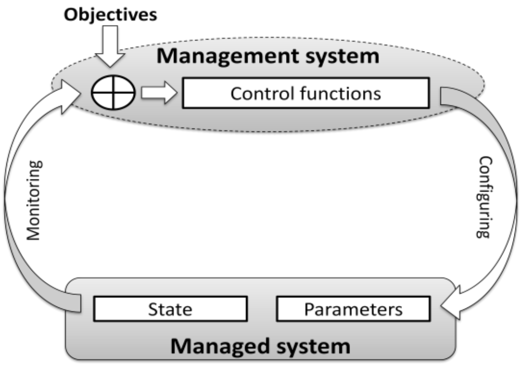


Figure 2.15: Management process

2.4.3. Networks management approaches for MANETs

In this section our attention is focused on managing ad hoc networks. Generally, network management applications include monitoring and configuration of network and node resources. However, management of ad hoc networks must be performed in a self-organized manner to ensure robustness and scalability. Unlike traditional fixed networks, MANETs rely on wireless connections between mobile nodes, which means high rate of disconnections between nodes, limited network and node resources. So, there is a great need for new management architecture able to cope with MANETs constraints.

By classical definition, “network management is a process of controlling a complex data network so as to maximize its efficiency and productivity” [47]. This process involves two main management activities, monitoring and configuring. As shown in Figure 2.15, the management process represents a function that takes as input both objectives, observer states, and eventually the old-settings configuration. It returns as output the proper configuration parameters applied to the managed system. Through this loop control, we can see that management process is similar to the concept of an automatic control system.

Designing network management system architecture is based on multidimensional modelling [48]. We consider three models that specify the conceptual aspect of information, organization, and communication in the management architectures. These models are not specific to ad hoc networking but are recurrent in network management.

- *Information model* defines a common formal framework for describing managed resources and the structuring of management information. A resource is typically modelled by an object their attributes indicate the state, and are accessible by management operations.

- *Organizational model* defines the role (manager or agent) of each component of the management plan and specifies how each component interacts with the others. Network management systems can be classified according to their network organizational model: centralized model, distributed model, hierarchical model
- *Communication model* specifies the protocol architecture to exchange management information between different entities. These include ensuring application-level exchanges between the manager and agent, in order to transmit operations and enable access and manipulation of data management.

Because of the nature of MANETs, these models need to be adapted. The role assignation and the relationships among components are highly dynamic in MANETs compared to regular fixed networks. After the maturity of ad hoc routing protocols, which dominated initial research interest, the need for managing ad hoc networks come out as an important issue and is receiving important interest. Among the specific issues that a MANET management solution must address are: QoS provisioning, network mobility, communication intermittence, node connectivity, network traffic, communication fairness, security and misbehaviour of communication and routing. We will present here three management approaches that have been proposed or redefined for the ad hoc networks.

2.4.3.1. SNMP-Based management

Current network management systems use a set of solutions. The Simple Network Management Protocol (SNMP) [49] is one of these solutions, which allows the exchange of management information between a network manager and the agents. SNMP is a widely common used protocol for fixed network management (monitoring and configuration). SNMP-Based approaches focus on developing a lightweight management protocol for MANETs that is compatible with SNMP protocol.

The Ad hoc Network Management Protocol (ANMP) [50] has been one of the first efforts by introducing an SNMP-based solution for MANETs. The compatibility with SNMP relies on using the same protocol data unit structure and the same management information base (MIB) structure. The major difference comes from the architectural model presented in figure 2.16, which combines clusters and hierarchical levels.

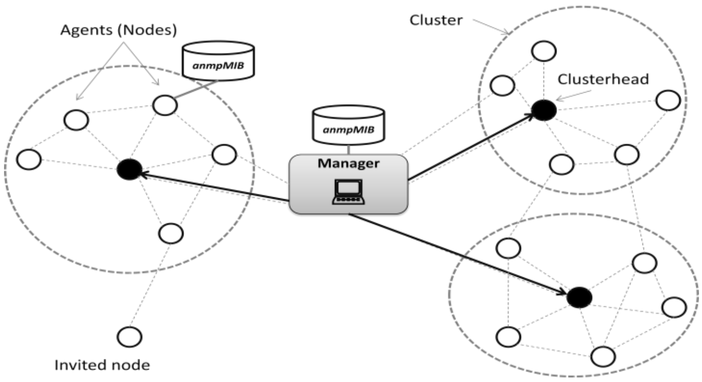


Figure 2.16: ANMP Architecture

ANMP comes within the scope of management by delegation. It clusters the management plane and organizes it in a three-level hierarchy composed of: individual managed nodes interacting as agents at the lowest level, cluster-heads interacting as local managers at the middle level and a central network manager at the top level. The aim of this hierarchical model is to reduce the network traffic overhead by collating management data at the intermediate levels. anmpMIB is a management information base defined by ANMP specific to MANETs as an additional group to the standardized MIB-II of SNMP.

An alternative SNMP-based solution is proposed in [51] by GUERRILLA Management Architecture (GMA). Uses the cluster based management mechanism and mobile agents in order to implement an autonomic management environment. Nodes are clustered into groups with at least one nomadic manager in each group. The nomadic managers collaborate autonomously to manage the entire network.

However, a problem in utilizing SNMP-based management architecture approaches in MANETs is the cost of maintaining a hierarchy (cluster construction and cluster head election process) or to disseminate requests and collect replies in the face of node mobility. Thus, introduce an additional overhead that will increase energy consumption and decrease the available bandwidth.

2.4.3.2. Policy-Based management

The traditional approaches to network management (such as the SNMP protocol) focus on individual devices and often rely upon proven technologies. However, it can be quite a time consuming resources if the number of managed nodes is great. The *Policy-based Network Management (PBNM)* [52] was originally defined by the IETF to manage the admission control and network resource reservation in Intserv context.

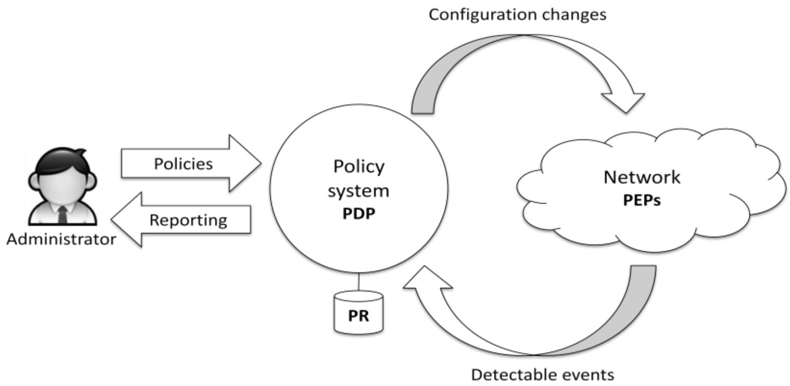


Figure 2.17: PBNM System Architecture

By using policies specification, PBNM simplifies the complex management tasks of large scale systems, since policies monitor the network and automatically enforce appropriate actions in the system. These policies specify the conditions that should be satisfied before executing management operations for any desired goals. Generally, policies are roles in the general form:

ON <event> **IF** <condition> **DO** <action>

As depicted in Figure 2.17 the policy-based management system defined by IETF has the following components: a Policy Enforcement Point (PEP), a Policy Decision Point (PDP), a Policy Repository (PR), and a Policy Management tool. The task of the PDP is to retrieve and interpret the policy information, and pass it to the PEP for execute the associate actions. The policy repository is the place where all policies are stored and from which they are taken by PDPs.

The Common Open Policy Service (COPS) [53] protocol was specified to standardize the distribution policy through the PDP-PEP communications. However, COPS protocol has found little acceptance and their relatively heavyweight nature has limited their applicability to MANETs.

DRAMA [54] is a pioneer work which has investigated PBNM system for MANETs, using intelligent agents. As illustrated in the Figure 2.18 below, Policy agents are deployed and manage the network through a two tier hierarchical architecture: Global policy agent (GPA) manages multiple Domain Policy Agents (DPAs). DPA can manage multiple DPAs or Local Policy Agents (LPAs). LPA manages a node. In order to reduce management bandwidth overhead, LPAs perform local policy-controlled configuration, monitoring, filtering, aggregation, and reporting. DRAMA uses Yelp Announcement Protocol (YAP) for efficiently reporting management information. Policies are distributed using a combination of DRCP/DCDP (Dynamic Configuration Distribution Protocol/ Dynamic and Rapid Configuration Protocol).

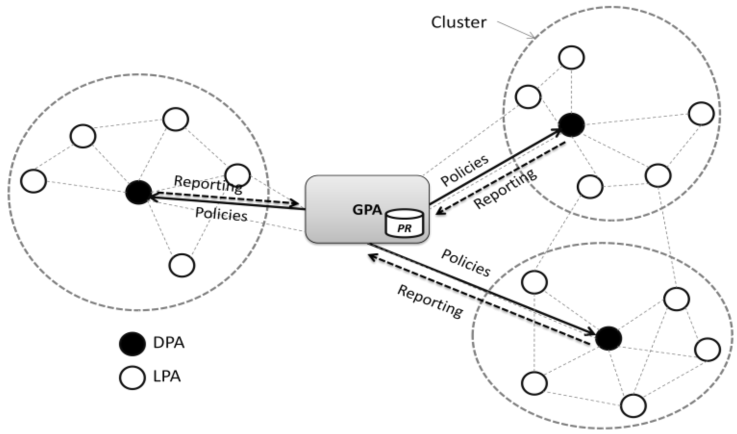


Figure 2.18: DRAMA Architecture

DRAMA management system was prototyped and demonstrated in a realistic environment to illustrate several use cases including CPU utilization reporting, server reallocation upon failure and reconfiguration of bandwidth allocation. However, the conceptual architecture of DRAMA is relatively simple, but allows an exhaustive set of experiments.

Table 2.7 summarizes the main characteristics of PBNM solutions for ad hoc networks presented in the literature. All these solutions are clustering based and differ in the communication patterns between different tiers. PBNM presents a better choice for managing dynamic networks, in particular QoS management. However, these policies are constrained by the limitations of IETF centralized architecture. It is better to move towards a distributed conception of PBNM. In this context, the self-management as defined in autonomic networks approach is a promising solution.

Solutions	**hierarchic**	**Agents-based**	**Protocols**	**Policies repository**
K. Phanse [53]	No	No	COPS-PR	PIB
DRAMA [54]	Yes	Yes	DRCP/ DCDP, AMPS, YAP	local data base (MySql)
Malatras [55]	Yes	No	XML-RPC	DPR (Distributed Policy Repository)

Table 2.7: PBNM solution comparison for MANET

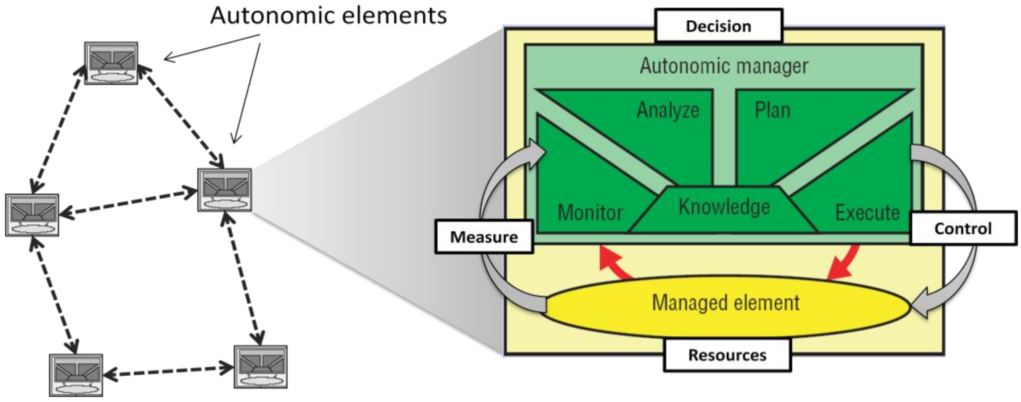


Figure 2.19: Autonomic Control Loop [56]

2.4.3.3. Self-management

The self-managing approach allows systems to manage themselves by given high-level objectives from administrators. This approach is based on autonomic computing paradigm that has been proposed by IBM in 2001 [56]. IBM has defined four general properties a system should have to constitute self-management:

- **self-configuring**: the ability of the system to configure and reconfigure automatically itself under varying conditions and changes in its environment,
- **self-optimizing**: the ability of the system to optimize efficiently its use of the resources by measuring and tuning them,
- **self-healing**: the ability of the system to detect instantly improper events and react to system disruptions,
- **self-protecting**: the ability of the system to detect, identify and protect against attacks.

Autonomic network architecture consists of a collection of autonomic elements. As depicted in Figure 2.19, each autonomic element is typically consisting of an autonomic manager and one or managed element. Managed element could be a resource such as storage, CPU, etc. Autonomic manager is composed of:

- Monitor: which is responsible for knowledge gathering,
- Knowledge base: which consists of a repository where policies and monitored information are stored,
- "Analyze and Plan" component: which analyzes knowledge and constructs plans of actions,
- Executor: which reconfigures the system regarding to the output of analysis and plan processes.

Autonomic computing seems very useful in the case of QoS management in MANETs, because these networks are self-creating and need to be self-managing. The application of the autonomic paradigms on MANETs is recent. In [57], an *Autonomous Decentralized Management Architecture* (ADMA) has been proposed. This solution improves the self-configuring capabilities in MANETs by totally distributing policy-based management system. ADMA gives the system the ability to configure and reconfigure itself under varying conditions and changes in its environment without a human intervention. ADMA has been implemented and evaluated under ns-2, and the performance of the designed solution was tested in the case of a real-time application (Voice over IP). The simulation results show an improvement of the VoIP delay and loss rate. This prior solution also has some drawbacks, in particular, when distribute policies over the network which generate an overhead and introduce latency.

Given the above overview, MANETs management architecture approaches have different limits. Hierarchical management model based on SNMP can perform complex management task and more accurate management decisions. Although, it has several inconvenient such as: high message overhead, single point failure, and in partitioned network some nodes left without any management functionality. Because of the self-organized characteristic of MANETs, the management task has to be distributed. Policy-based network management offers relatively this feature, by executing and applying policies previously defined by network manager. Otherwise, the complexity of realization and control makes its realization rather difficult. Autonomic network trend proposes the self-management solution. This approach leads to network management architecture able to take into consideration the autonomous nature of MANETs. A drawback of the self-management approach is that it may be too computationally for resource constrained nodes and it needs sensor nodes to be dedicated for specific management roles.

Messages overhead generated by the management system is one of the crucial issues that the network management architecture designers must deal with. Knowing that, management system should not be hyper-consumer of network and nodes resources, especially in MANET environment. Therefore, exploiting the routing plan for providing information and communication to the management system could be very useful in this case. Routing protocol (e.g. OLSR, AODV) can be utilized to report monitoring management information or configuration parameter, these can reduce management message overhead and enhance the performance of MANET. This approach will be investigated in this work.

In the next section, we explore Voice over IP in MANETs context. From these, we provide backgrounds and a literature review for VoIP application over MANETs. Additionally, we discuss voice coding adaptation mechanism.

2.5. VoIP over MANETs (VoMAN)

2.5.1. Challenges

One application domain for ad hoc networks is to support real-time traffic and, in particular, voice. However, MANET as most of the IP-related networks was not designed having voice in mind. As such it brings a lot of new limitations and introduces many challenges for successfully deploying VoIP on top of it. These challenges include:

- Multi-hop routing and dynamic route calculation.
- Wireless channel: Inherent broadcast nature and temporal response variability due to fading, absorption and noise- and interference-sensitivity.
- Scarcity of resources
- Lack of a central entity
- Volatility of connections

Two major factors associated with packet networks that have a significant influence on perceived speech quality are packet loss and delay. In addition to MANET constraints, voice quality is mainly governed by VoIP packets which usually have other characteristics which demand a very well-configured network to run smoothly. These characteristics include:

- Small VoIP packets and short packets intervals: interval times for the sending of voice packets are shorter than the ones used for data packets, usually 10-30*ms*, and the packets have a very small payload of between 20-160 bytes based on different codecs that are used.

- Header to payload is relatively high: for transmitting 20-160 bytes a header of 40 bytes is needed. Thus, the transmission of VoIP packets may not be an efficient process, due to the high ratio between the header and payload size,

Moreover, the choice of codec also has a substantial impact. For example, the number of codec samples per packet is a factor that determines the bandwidth and delay of a VoIP call. The codec defines the size of the sample, but the total number of samples placed in a packet affects how many packets are sent per second. If voice payload size is increased the VoIP bandwidth reduces and the overall delay increases.

When combined, these challenges make supporting real-time applications over MANETs a good task. Therefore, VoMAN issues are gaining a special attention from the research community trying to create new assessment and management paradigms. In the next subsection, we review works related to voice transmission over WLAN, and over MANETs.

2.5.2. Related work and literature review

Significant works in the area of VoIP over WLANs exist, whereas the literature on VoIP in MANETs is very limited.

Various studies consider voice capacity of WLANs. In [58] and [59] Garg and Kappes use both an experimental and an analytical method to calculate the upper bound of the number of VoIP calls in a cell. This upper bound turns out to be very low, with only 6 calls supported using a G.711 codec and a 10*ms* packetization interval (12 calls with a packetization interval of 20*ms*). A number of further studies like the ones by Hole and Tobagi [60] and Wang et al [61] validate these results. In [72], where several voice codecs have been studied over WLAN, the authors found through simulations that G.723.1 has the ability to provide the highest capacity for VoIP calls.

Several works have been focused on voice support over ad hoc networks without considering mobility (e.i. Wirsless Mesh Ntworks (WMNs)). In [62], the authors analyzed end-to-end performance enhancements by introducing approach based on aggregation. In particular, author proposes concatenation algorithms to reduce VoIP protocol overhead in a real multi-hop mesh network. The main benefit of these mechanisms is an increase in the number of calls that can be supported in the mesh network; however, the voice quality resulted to be affected by the aggregation. In [63], the authors made extensive experiments on a real WMN that supports VoIP. They have analyzed the performance in terms of delay and jitter at the increase of the service traffic load in the network. The result is that VoIP performs flawlessly over up to five hops, whereas some degradation are observed when there are any other types of traffic flows in parallel. In [64], the authors have analyzed the performance of VoIP systems in ad hoc networks with stationery nodes when using two routing protocols: AODV and OLSR. Also the authors in [65] have investigated the deployment of VoIP services with the AODV routing protocol and analyzed different performances metrics, such as jitter, one-way delay. Both papers have highlighted that the available routing algorithms still need to be improved to support telephony services over mobile ad hoc networks at a satisfactory quality. In [66], authors evaluate the performance of different reactive routing protocols, such as the AODV, DSR and TORA, when varying the load of real-time traffic. These routing protocols introduce high transmission delays because they determine the routing table only if there is traffic in the network. A modification of IEEE 802.11 is proposed in Dong et al. [67] in which the cyclic redundancy codes are computed only over those parts of the voice frame that have a high impact on the perceived quality rather than over the entire frame. In this way, less bandwidth is wasted in retransmission and less delay is introduced.

Only few works appeared in the last years on the analysis of quality issues in real-time traffic transmission over MANET networks. In [68], the simulations for VoIP and Video traffic simulation over mobile tele-emergency system were investigated; the authors conclude that G.723.1 worked well in both small and medium scale network. The authors in [69] show how even a single VoIP flow traversing 6 wireless hops can completely occupy the bandwidth in a wireless network. The number of hops plays an important role in degrading the voice quality since more hops imply a higher loss probability. In [70], the use of new speech coding techniques for supporting voice over ad hoc networks is proposed. One such technique is multiple description coding. It involves creating more than one bit stream from the source signal. Each independent stream represents a coarse description of the transmitted signal. If more than one description is received, a refined signal is reconstructed. Our work does not make use of these speech coding techniques.

Call Admission Control (CAC) algorithms are components of a system attempting to provide QoS, particularly in wireless mobile networks. CAC algorithms work by regulating the total number of calls passing a specific point per unit time. If a defined limit is reached or exceeded, a new call may be prohibited from entering the network until at least one current call terminates. An extensive survey of the common CAC techniques in WLANs can be found in [71]. The research on CAC methods is wide and much effort is placed on controlling voice QoS on a wireless cell and distributes the resources of it efficiently. However, none of these methods are actually dealing with the QoS of the voice flows once accepted at the network, or with the QoS degradation provoked by the capacity variable channel to the already active calls.

Link Adaptation mechanism (multi-rate channel) is one of the most important factors that affect the voice call. Heusse et al. [72] identified an anomaly caused by this mechanism. They observed that the rate change of one of the nodes of the cell causes a general degradation on the transmission rate of all other nodes. In VoIP scenarios, several studies have been shown that adjustment of VoIP codec rate based on network conditions maintains an efficient utilization of the available resources as a consequence increases QoS when using. A number of works consider adaptive codec selection method over multi-rate channel. Adaptive modulation and adaptive compression have been applied separately in VoIP-based wired [69-73] and WLAN, [74-75] networks. Also, are investigated in MANETs. Fasolo et al. [76] present a cloud of nodes that communicate with one gateway by means of multi-hop ad hoc connections to study the effect of multi-rate on voice capacity. The authors states that; minimizing the length of the paths and the number of overlapping paths is needed to increase the system capacity. Zhang et al. [77] propose an adaptive coding control scheme to adapt the voice coding bit-rate to the available network bandwidth. However, the results have been compared with CBR scheme which not reflect effectiveness of the proposed solution. Also, in the next section we review briefly two other codec adaptation proposals concerning respectively WLAN [78] and MANETs [79].

2.5.3. Codec adaptation proposals

2.5.3.1. Link adaptation over multi-rate wireless channel

The wireless channel characteristics are very dynamic in nature, which is further aggravated due to mobility. Multi-rate transmission capabilities provided by PHY/MAC specifications of the IEEE 802.11x standards (Table 2.8) allow each mobile node to adapt their transmission rate dynamically by selecting its physical layer parameters (modulation and channel coding). This mechanism occurs on the mobile nodes under varying conditions (user movement, meteorological conditions, interferences etc) to optimize the bit transmission over the noise/fading-prone channel. Additionally, the use of multi-rate PHY permits the use of the different communications range dependent on channel condition. Figure 2.20 illustrates an example of 802.11b Standard which operates under different rate (1, 2, 5.5, 11Mibt/s). Thus, the selection of a modulation and a codification rate satisfies.

802.11x standards	Rates (Mbits/s)
a	6, 9, 12, 18, 24, 36, 48, 54
b	1, 2, 5.5, 11
g	6, 9, 12, 18, 24, 36, 48, 54
n	7.2, 14.4, 21.7, 28.9, 43.3, 57.8, 65, 72.2
p	3, 4.5, 6, 9, 12, 18, 24, 27

Table 2.8: Offred rates of IEEE 802.11x standards

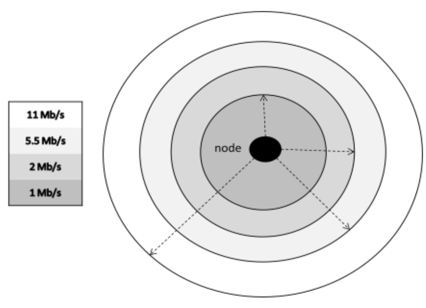


Figure 2.20: Transmission rate depending on transmission range (802.11b)

However, multi-rate has a direct impact on voice transmissions, and produces a general degradation of the QoS and a very hostile environment for VoIP. The main reason for this is that by reducing its transmission rate a node demands more network resources (channel occupancy time) in order to transmit the same amount of data as before. This increased demand can lead the system to a congestion state with a direct impact on all active calls. Hence, the codec bite-rates can be dynamically selected with respect to channel data-rate.

To show the relationship between actual required bandwidth and speech coding rate, we investigate the capacity of a wireless single hop ad hoc network in supporting the packet voice. Single hop ad hoc networks are based on CSMA/CA using DCF 802.11. One may note that the capacity of the channel is affected by various protocol overheads. According to [77], the actual required bandwidth B for a speech call is:

$$B = r + \frac{H}{T} + \frac{T_{sr}R}{T}$$

Where:

$$T_{sr} = DIFS + T_{slot}\,(CW_{min} - 1)/2 + PHY + SIFS + ACK$$

And

H: packet header,
r: speech coding rate,
T: frame length in terms of time,
R: data transmission data rate.

From the above formula we know that, channel data rate and speech coding rate have a direct impact on the required bandwidth. So choosing a good quality, high rate voice coder (e.g., G.711 at 64 kbps) may not be possible at all times given bandwidth scarcity. Otherwise, a very low rate coder (e.g., G.723.1 at 5.3 kbps) may be used to curb the chances of overloading the network. So, the use of voice coding adaptation mechanism is needed. Next we present work related to codec adaptation in WLANs forwarded by other in ad hoc networks.

2.5.3.2. Codec adaptation over WLAN

Sfairopoulou et al. [78] study the effect of multi-rate WLAN networks on VoIP traffic. As a result, a cross-layer algorithm is proposed that both monitors the QoS of the calls and adapts their voice codec when considered necessary. This cross-layer codec adaptation uses the advantage of combining information distributed between different layers in the 802.11 architecture and the SIP mechanism for session renegotiation.

The codec adaptation approach is based on the simple concept that different codecs have different bandwidth needs. Thus, in order to lower the total relative bandwidth demands of the VoIP calls, a lower bit-rate codec can be used in some of them, which at the same time can preserve the QoS at acceptable levels. Consider a

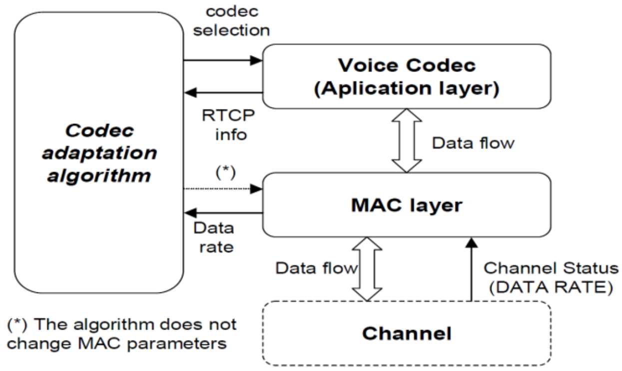


Figure 2. 21: Information flow of the codec adaptation solution [78]

scenario, with a number of VoIP calls active. When a node changes its rate to a lower one, this will most probably have a direct effect in the perceived QoS of all the VoIP flows in the cell. Therefore, a monitor is needed so as to capture these rate changes. In case that a rate is changed, a change on the codec of some of the active calls would help to lower the congestion level of the cell and recover the QoS levels. Moreover, a RTCP filtering function is used for real-time feedback on the quality of the VoIP flows, such as delay, jitter and packet loss. The basic structure of the codec adaptation over multi-tare WLAN proposal can be seen in Figure 2.21.

The results have shown an increase of the WLAN performance in terms of higher capacity (successful calls) and perceived speech quality. However, this solution wasted network resources because the RTCP control packets. Knowing that MANET has already a routing control overhead, using RTCP protocol to report the QoS state may add additional overhead which make the bandwidth extremely limited.

2.5.3.3. Opportunistic adaptation over MANET

Obeidat et al. [79] propose an opportunistic adaptive protocol within a cross-layer framework to support VoIP over multi-hop wireless networks by combining the merits of both adaptation and cross-layer design. In order to maximize the number of calls admitted while minimizing loss of quality. Authors propose an opportunistic protocol within a cross-layer framework that adapts factors (modulation, compression, packet size) influencing voice communication, at different time scales. The performance of the protocol was evaluated through simulations in static and mobile scenarios, carrying real-time audio traffic using both quantitative and qualitative audio metrics. As depicted in Figure 2.22, adaptive protocol architecture is based on two adaptation mechanisms: Hop-by-hop adaptation and end-to-end adaptation.

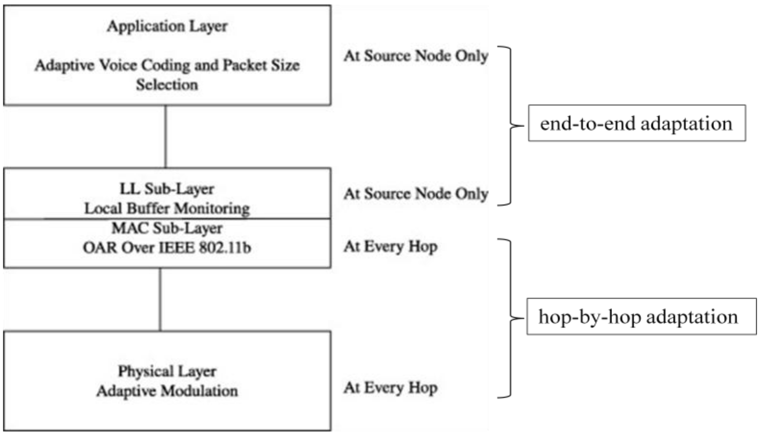


Figure 2.22: System architecture [79]

Hop-by-hop adaptation exploits the physical and MAC layers interaction to improve the use of the spectral resources through opportunistic rate-control and packet bursts. Cross communication between the PHY/MAC layers takes place at every hop along the path from the source to the destination and enables adaptive modulation using the *opportunistic auto rate* (OAR) protocol over IEEE 802.11b to make use of the multi-rate capability of the PHY layer. End-to-end adaptation takes place only at the source and exploits the logical link control and application (LLC/APP) layers interaction to control the demand per call through voice coding and packet size selection. Compression and packet size selection depend on the end-to-end feedback regarding the network conditions expressed in terms of the packet loss ratio and average packet delay.

The performance of the solution was evaluated through simulation using qualitative and quantitative metrics. However, this solution must be evaluated under different routing schemes in order to show the routing layer impact. Additionally, the use of the large-scale adaptation may generate delays spatially in high dynamic scenarios.

2.6. Conclusion

In this chapter, we presented background information related to main building blocks of our work. Firstly, we reviewed the VoIP elements, starting from VoIP system to the most known quality of service measurement tools. Then, we introduced the mobile ad hoc networks, its application, characteristics, and routing protocols. Existent solutions for QoS requirement and management in MANETs have been presented and discussed. In the last section, we presented works related to VoIP support over MANETs focusing on adaptation mechanisms solutions.

Despite the great number of proposals, providing quality of service for VoIP over MANETs is still an open issue. According to the literature review performed in the last section, we observe that most simulation works addressing VoIP performance over MANETs don't consider a VoIP system which reflects a real implementation, such as using CBR traffic to simulate voice traffic. Furthermore, the interaction between VoIP application and other network protocols has never been investigated.

In the next chapter, we provide a comprehensive simulation studies which will guide us in the direction of developing a solution for the enhancement of VoIP transmission support in MANET.

Chapter 3

VoIP traffic over MANETs: Comprehensive studies

3.1. Introduction

In order to understand the MANET behaviour when transmitting real-time traffic (VoIP), this chapter provides comprehensive studies concerning series of simulation in ns-2 [99] relevant to different VoMAN scenarios. The first Section describes the methodology and tools used to carry out the experiments reported in this dissertation. In Section 2, we determine whether routing protocols affect VoIP performance. For this task, a performance comparison of two routing protocols (proactive and reactive) is performed. In Section 3, a performance evaluation of various voice codecs and its impact on quality of service metrics will be analyzed. In last Section, we take the first step in addressing the practical issues related to the impact of routing parameter on VoIP codec. The approach is based on tuning these parameters (refresh interval and willingness) and measuring QoS metrics.

3.2. Experimental methodology

3.2.1. Performance evaluation approach

Performance evaluation determines performance measures for existing systems or, for models of (existing or future) systems [100]. Generally, the performance evaluation can use one of the following approaches:

Measurement: this technique is only possible if system prototype already exists. It can be done in a testbed network or an operational network. Measurement requires real equipment, code and time to run experiments (monitoring, prototyping). Up till now, although several MANET testbeds have been built [102], there are still no widely accepted measurement tools or methods that prove the efficiency of MANET protocols. However there might still be difficulties in carrying out the performance measurements because of cost and security/privacy issues. It may be too disruptive to do a measurement by testing on real operational networks.

Analytical model: uses mathematical notions and models to describe performance aspects of the system under study. It requires construction of a mathematical model of the system such as using queuing networks and Petri-Nets. Analytical modelling results can have better predictive values than measurement or simulation. However, an analytical model generally requires too many simplifications and assumptions, which may be inaccurate about the real network. Thus, most network protocols and systems are too complex to be realistically modelled using analytical modelling.

Simulation: One of the most commonly used paradigms in the study of communication networks. There are a number of credible published research works has been done using network simulation (e.g. those that appear in IEEE/ACM journals and proceedings). Simulation-based studies permit to ensure that functional requirements of new network elements (e.g. new protocols) can be achieved and are feasible. It offers on one side a flexible and scalable approach to create reasonably detailed physical and link models, and to repeat and control target network conditions. On the other side, a convenient approach in varying network parameters such as bandwidth, latencies, error rates, workload and system scale to better understand the performance under a wider variety of conditions, than may be possible in real networks with real implementations. One potential problem of simulation-based study is that simulations may take a long time. Therefore, with simulations, it is critical to search the space of parameter values for the optimal combination and perform factorial analysis, to identify the impacts of various factors.

Simulation-based performance evaluation can be performed using network simulation packages/tools. Minimally, a network simulator should enable users to represent a network topology, defining the scenarios, specifying the nodes on the network, the links between those nodes and the traffic between the nodes. Table 3.1 describes some of popular network simulator tools.

License	Simulator	Characteristic	Language	Recent release
Open source	Ns-2	OOP, discrete event	OTcl/C++	35
	Ns-3	discrete-event	C++/ Python	16
	Omnet++	component-based	C++	4.3
	Glomosim	parallel programming	C	2.03
	Jsim	Java-based	Java	2.10
Commercial	Qualnet	GloMoSim based		5.1
	Opnet			

Table 3. 1: Most used network simulator

Considering the unavailability of *real* MANET systems, we use the simulation approach to estimate and approximate the real performance in this study. We have chosen the discreet event Network Simulator 2 (ns-2) [99] to carry out different simulations, because it presents several features which make it suitable for our research; these features include:

- Cheap: does not require costly equipment
- Open-source: ns-2 is fully free to use. With the source code we could verify the correctness of the implementations and debug the source of possible errors.
- Support ad hoc: provides the most comprehensive implementations, especially for MANET routing protocols.
- Popular: has the largest user community; it is therefore much easier to get help and support from online FAQs/tutorials and other online communities such as mail lists.
- Extensible: The object-oriented software architecture of ns-2 makes it easy to add new modules and extend existing ones using object inheritance. However, adding new components is a hard task since it requires a good knowledge of the simulator.

We provide more details about ns-2 in Appendix A. However, ns-2 lacks a sound and flexible simulation model of a VoIP application. Note that in most performance studies made on VoIP application in MANETs, VoIP traffic is presented as a special case of CBR traffic, which does not reflect the reality. Thanks to the contribution of Andreozzi et al. [103], a module for simulating realistic VoIP traffic with ns-2 was developed and we have adopted it for our research. In the next section, we give an overall overview concerning this module.

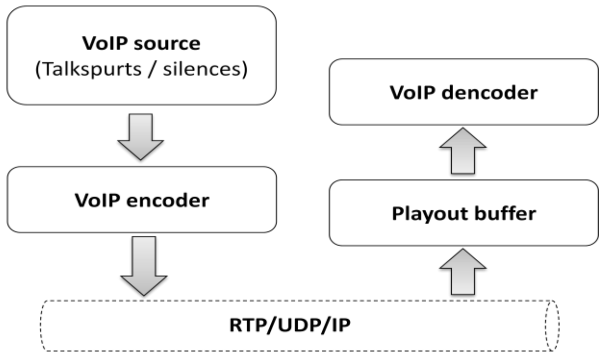


Figure 3.1: VoIP module structure [103]

3.2.2. VoIP traffic simulation under ns-2

As mentioned, to simulate voice traffic, we have reused *ns2voip++* framework developed by Andreozzi et al. in Pisa University [103]. This module allows simulating VoIP sources and destinations, and implements principal functions for compression, decompression and aggregation of voice traffic.

The structure of the VoIP module is illustrated in Figure 3.1. A VoIP source models the user activity, deciding whether a user is in a talkspurt or in a silence period. In the former case, the source signals to the encoder the beginning and the (randomly selected) talkspurt duration. During a talkspurt, the encoder generates data according to the chosen codec. The latter defines the size of each packet and the frequency of their generation. At the receiver side, a VoIP decoder receives packets from the network and employs a playout buffer to pace their playout. The MOS, as well as other performance metrics, are evaluated at the decoder. More detail about *ns2voip++* is provided in Appendix B.

In *ns2voip++* four VoIP codecs are implemented (e.g., G.711 and GSM.AMR). We have added two other codecs (G.726, G.728). These codecs are characterized by:

- *Bitrate* which is the number of bits per second required to deliver a voice call,
- *Sample size* which is the number of bytes captured from the analog signal during the *sample interval,*
- *Sample interval* represents the time between samples.
- *Payload size* is the bytes that fill the packet, depending on the codec and the header size required for coding, packets can have various payload sizes. However, the payload size must be a multiple of the sample size [1].

Codec	Sample size (bytes)	Sample interval (ms)	Bit-rate (kbps)	Payload size (bytes)	MOS (score)	Ie (%)
G.711	80	10	64	160	4.1	0
G.726	20	5	32	80	3.85	7
G.728	10	5	16	60	3.61	7
G.729A	10	10	8	20	3.7	11
G.723.1	20	30	5.3	20	3.65	19

Table 3.2: Description of voice codecs investigated in this work

Codec taken into account in our study are illustrated in Table 3.2. We chose these category standards because they present overall codec characteristics and are among of the codecs commonly used today in VoIP application.

Each codec is given a theorecal MOS value based on the ITU-T *impairment factor* (*Ie*) [104] which is a percentage value that characterizes the effect of coding on quality. The higher the *Ie*, the higher the impact on perceived quality [19-104]. Also, the higher the compression (bit-rate), the higher the penalty of packet loss in terms of the effect on perceived quality. In this study, we use **Ns2mesure** framework [105] for MOS calculation. This tool is a set of routines that allow E-model statistics to be gathered and computed directly as an outcome of simulation runs.

3.2.3. Data collection, processing and analysis

3.2.3.1. Data collection

There are many ways of collecting output or trace data on a simulation. Generally, trace data is either displayed directly during execution of the simulation, or (more commonly) stored in a file to be post-processed and analyzed. There are two primary but distinct types of monitoring capabilities currently supported by the simulator.

- Traces method: records each individual packet as it arrives, departs, or is dropped at a link or a queue. Trace objects are configured into a simulation as nodes in the network topology.
- Monitors method: records counts of various quantities such as packet and byte arrivals, departures, etc. Monitors are supported by a separate set of objects that are created and inserted into the network topology around queues. They provide an easy way to calculate arrival statistics.

```
r -t 0.004655830 -Hs 25 -Hd -1 -Ni 25 -Nx 805.00 -Ny 1272.00 -Nz 0.00 -Ne -1.000000 -Nl RTR -Nw --- -Ma 0
-Md fffffff -Ms 4 -Mt 800 -Is 4.255 -Id -1.255 -It OLSR -Il 48 -If 0 -Ii 0 -Iv 32 -P olsr -Pn 1 -Ps 0
[-Pt HELLO -Po 4 -Ph 0 -Pms 0]
```

Figure 3.2: Standard trace example

Compared with monitors, traces are more generic and extensible. It is easier to use traces to implement customized data collection methods. Thus, in our study, we use traces to record network events and perform post-processing. Ns-2 currently uses *cmu-trace* [106] (the default implementation of wireless extension) support for wireless simulations objects. When running the Tcl simulation script, the trace result will be stored in trace file (for example: "out.tr"). The trace file format defines how the variable details are recorded in a trace file. Figure 3.2 presents an example trace item from our simulation trace which leads us to obtain the following information: A HELLO message of OLSR agent is broadcasted by node 0 25 at time 0.004655830; the packet length is 48 bytes; it is a routing control packet, instead of data packet.

3.2.3.2. Data processing

The large volume of trace files makes trace based data analysis a problem. For example: The size of a typical trace file, generated by one single high-density network simulation run (for example, a network with 50 nodes), is around 200 Mbytes. Therefore, considering the large cost in storage, the trace files are processed immediately after each simulation by using extensible data processing tools as a toolkit. In our study we use **AWK Programming** to extract performance information from our trace files by given each ns-2 trace file as input.

AWK is a flexible UNIX text manipulation utility. Its name is derived from the family names of its authors (Aho, Weinberger and Kernighan). A trace file is treated as a sequence of records, and by default each line is a record. Each line is broken up into a sequence of fields (`$1, $2, $3,…, $n`), so we can think of the first word in a line as the first field, the second word as the second field, and so on. An AWK program is of a sequence of pattern-action statements. AWK reads the input a line at a time.

To automate the whole simulation process we use **shell scripting** which is a UNIX (bash) command language interpreter. Shell scripts execute ns-2 to generate multiple scenarios which generate trace files, and then call the trace-analysing functions in the AWK scripts to analyse the trace data. Finally, the file that contains the required information is being represented by a plotting program (such as **Gnuplot**), the output is gotten in the form of graphs.

3.2.4. Simulation scenarios description

In our simulations, we use two types of MANET scenarios: *synthetic scenarios* and *concrete scenarios*.

Synthetic MANET scenarios provide a range of comprehensive simulation environments, which enable us to carry out performance evaluation by subjecting the protocols under consideration to more stringent network conditions such as higher mobility and higher density, higher load, etc. Major MANET simulation scenarios in this work are synthetic, where parameters such as mobility, traffic load and hop count are varied over an arbitrary range of values.

The concrete scenarios reflect specific MANET applications (e.g. urban scenario, rescue scenario, conference scenario, etc.). Results obtained from concrete scenarios are considered more realistic in terms of real-world deployment. We define a concrete simulation scenario in Subsection 4.3.2, namely *urban VANET scenario*. This scenario represents a VANET application deployed in restricted areas with obstacles, such as constrained road topology, multi-path fading and roadside obstacles, varying vehicular speed and mobility, traffic lights, traffic congestion, etc.

Generally, each performance study has to define different factors which impact the performance metrics. This research concentrates on three factors:

- Voice codec: refers to codec parameter used in VoIP communication, including bit-rate, sample interval, and payload size of the VoIP packet being sent. The basic parameters considered for the set of voice codecs used are shown in Table 3.2.
- VoIP calls: concern the number of VoIP connection established. We note that traffic differentiation and prioritization capabilities are not considered since there is only voice traffic. The inclusion of elastic or other types of rigid flows will be considered in further work.
- Routing protocol parameters: include the refresh intervals of routing protocols (i.e. HELLO interval and TC interval for OLSR) and other parameters.

The flow chart in Figure 3.3 illustrates our simulation process for different scenarios and conditions. The main source file is simulated with a specific scenario and factors. Then, the QoS metrics measurements are sieved out from trace output file. Finally, Graphs are drawn based on the average values from the simulation output files. Note that for each scenario, we generate 10 to 20 different instances having the same simulation parameters and characteristics. Only simulation *seeds* which are varied each time generating different instances (i.e. nodes speed and position). The results shown in the following sections represent the average results for the10 to 20 generated instances.

In this section, we described briefly the design of our experimental studies. In next section we will start our experiments by simulating VoIP in MANET context.

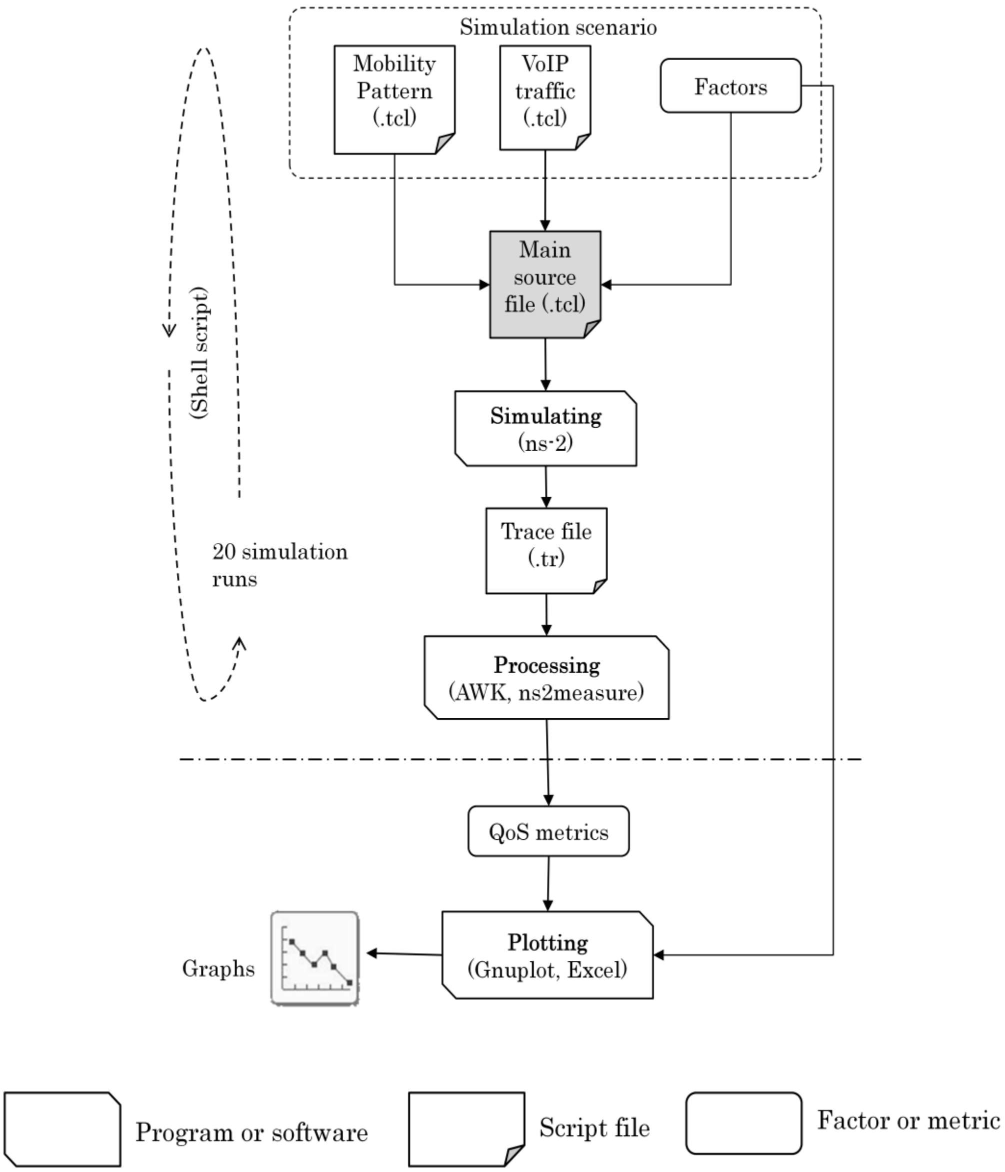


Figure 3.3: Simulation process

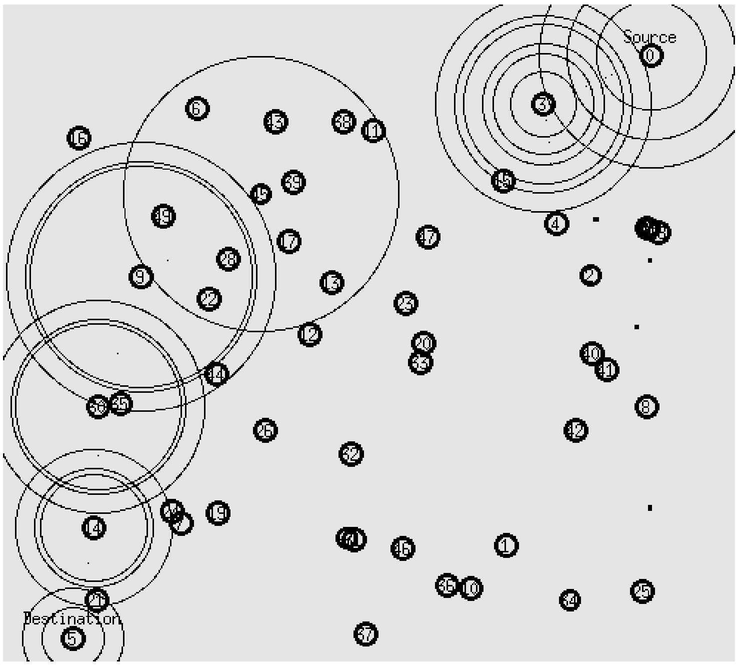


Figure 3.4: Simulation topology

3.3. Impact of routing on VoIP performance

Multi-hop environments create situations not normally seen in infrastructure-based one. The goal of this research is to determine whether routing protocols affect VoIP quality. This goal is met by investigating the performances comparison between the proactive and reactive routing protocols (OLSR and AODV) in a MANET carrying VoIP traffic. Via a simulation study, we examine whether hops affect VoIP performance and evaluate QoS indicators such as end-to-end delay (E2E-delay), packet loss and jitter.

3.3.1. Simulation scenario

Measurement of an actual MANET is expensive and infeasible. Therefore, the evaluation technique is simulation; we have used the discrete event network simulator ns-2.34 [99]. As shown in Figure 3.4 the simulation scenario consists of 50 nodes in an area of 1000x1000m created with random placement. Here, we are trying to transmit voice data over wireless multi-hop network, and testing how hops affect the quality of the voice.

To simulate and analyze performance of OLSR protocol, UM-OLSR [107] implementation is installed too. We use DCF IEEE 802.11 for the MAC layer. Finally, at the application level we used VoIP communication generated via *ns2voip++* module which is an extensible VoIP application implemented in ns-2 (see

Subsection 3.2.2). One VoIP streams is sent across the network from node source to node destination with variable duration. The VoIP source is configured to draw the duration of the talk-spurt and silence periods from *Weibull* distribution.

The overall system performance was tested using the average results obtained from 20×2×2 MANET scenarios defined in the specification presented in Table 4.3. The analysis focuses on the results considering static and dynamic scenarios. Two codecs are used; one with high bit-rate (G.711) and other with low bit-rate (G.723.1) over 100 seconds of simulation time.

Parameters	**Value**
Routing protocol	OLSR/AODV
Mac/Phy	DCF 802.11
Area	1000m^2
Traffic type	VoIP
Codecs	G.711/G.723.1
Number of nodes	50
Nodes position	Random
Mobility model	RWP (dynamic scenario)
Queue type	/DropTail/PriQueue
Simulation time	100s

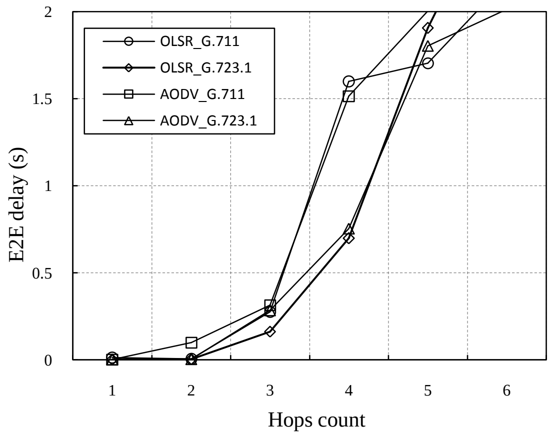
Table 3.3: Simulation parameters

3.3.2. Results and analysis

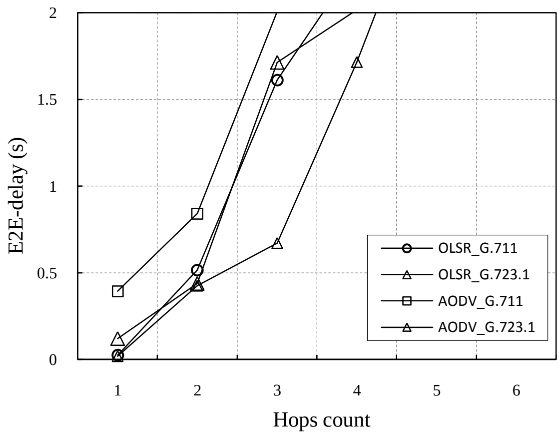
In the following, we show and discuss our simulation results investigating the impact of routing mechanisms in the both proactive and reactive context. QoS will be measured in terms of E2E-delay, packet loss, and jitter.

3.3.2.1. E2E-delay

Figures 3.5 (a) and (b) show the obtained E2E-delay for the considered routing protocols. As depicted in these figures delay is the lowest when hops are less than three, as the hops number increase, delay increases and exceeds the threshold recommended (*150 ms)*. However, the reactive protocol AODV presents more delay than the proactive protocol OLSR in dynamic scenario. This is due to the reactive AODV route maintenance generating an increase in transmission queues when the topology changes. This validates the choice of proactive routing protocol for delay-sensitive applications in several works [64-85-86].



(a)

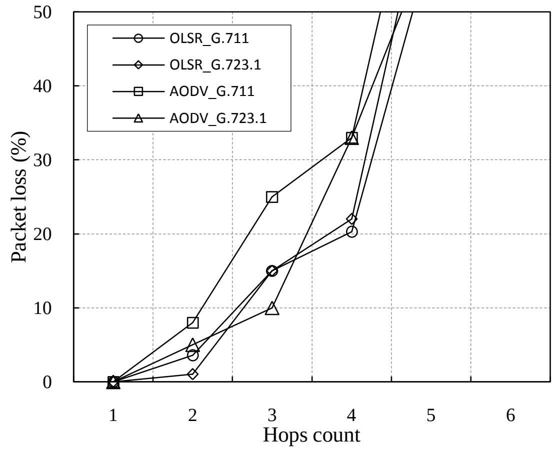


(b)

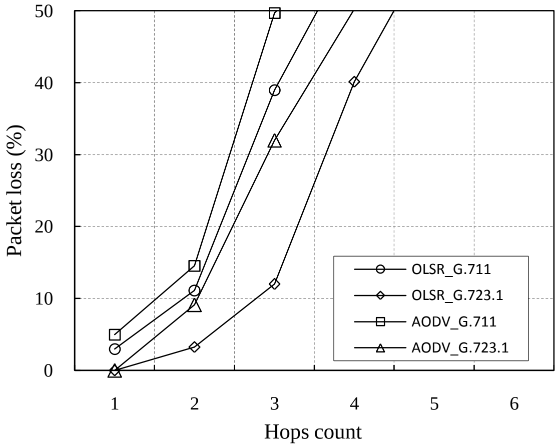
Figure 3.5: E2E-delay of the VoIP flow in static (a) and dynamic (b) scenarios

3.3.2.2. Packet loss

Packet loss is expressed as a ratio of the number of packets lost to the total number of VoIP packets transmitted. Packet losses results when packets sent are not received at the final destination. As shown in Figures 3.6 (a) and (b), the percentage of packet loss increases for different coding technique at three hops count (two for OLSR). For the simulation analysis, AODV suffers dramatically from the packet loss compare to OLSR. Generally, packet loss is related with the packet length, which is proportional to transmission time associated with each packet. Furthermore, the time intervals between packets are shorter in G.711, which worsens the performance in terms of dropped packets for both protocols.



(a)



(b)

Figure 3.6: VoIP packet loss in static (a) and dynamic (b) scenarios

3.3.2.3. Jitter

Delay variation (jitter) represents the relative delay between successive VoIP packets arriving at destination. It is a measure of delay variance. Ideally, jitter should be less than 57*ms* and each packet must take equal time in travelling from source to destination. In this scenario, we don't implement any de-jittering buffer mechanism. The jitter is plotted with respect to the number hops count (Figures 3.6 (a) and (b)).The behaviour of curves depicts that with an increase in number of the hops jitter increases and it becomes unacceptable with number of 4 hops in static environment and 3 hops in dynamic environment. Generally, OLSR has low values of jitter in regard to AODV which show that it has got a good behaviour.

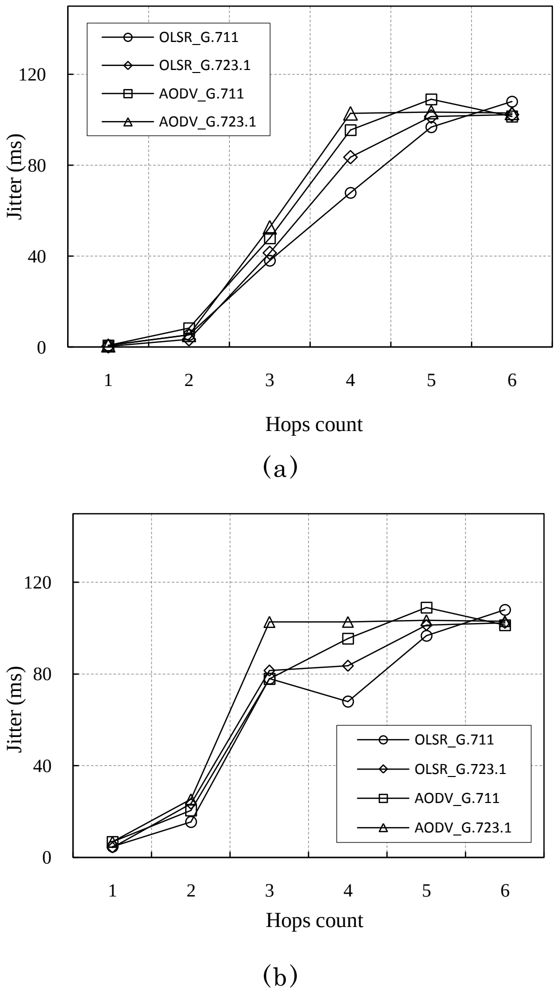


Figure 3.7: jitter in static (a) and dynamic (b) scenarios

In VoIP transmission the problems of simultaneous packet transmissions are amplified. A possible solution to these problems is to use the playout buffer. In the next simulations we use the playout buffer of *ns2voip++* framework in order to control packet transmissions and eliminate the jitter,

3.3.3. Synthesis

The hypothesis is that routing protocols have a significant role in VoIP performance in a MANET. Since the route calculation strategies of the two protocols are very different, in this section we investigate how VoIP application is influenced by wireless multi-hop network characteristics in order to optimize it for providing scalable communication. Considering the QoS requirements of a VoIP application, OLSR always presents an adequate behaviour in end-to end delay since valid routes are determined in advance. Based on this, OLSR had shown the best initial performance compared to AODV. Hence, we propose it for such real-time VoIP conversation applications in an ad hoc network scenario. Objective measurement of packet loss and jitter determine whether OLSR provides acceptable performance on the MANET. This research determines the suitability of OLSR as a routing protocol for MANETs running a VoIP application.

3.4. Voice codec behaviour in MANETs

More specifically, the target of this section is to study the performance of VoMAN (Voice over MANET) transmission when using different codecs. For this reason, we perform number of MANET simulations under different conditions. The approach is based on voice streaming between nodes rely on multi-hop and evaluating the performance of various voice codecs and its impact on quality of service metrics.

According to a definite previous study the proactive OLSR routing protocol is more suitable for application scenarios that support VoIP applications. Hence, we use in this simulation. Our simulation study is based on two processes:

- Generating several MANET instances (node position, node movement,) to achieve accurate simulations.
- Studying different QoS metrics presented in terms of both network level (such as E2E delay) and user level (MOS) metrics.

3.4.1. Simulation scenario

The considered scenarios focus on the unicast transmission of voice signals between multiple nodes. In the experiment carried out in this study, IEEE 802.11b standard is used with *TwoRayGround* propagation model. Table 3.5 summarizes some important features of the network used in our simulations. In order to analyze how various voice codec behave under such MANET environment, two scenarios are considered. The number of voice call is dependent of simulation scenario and are performed during different moments of the simulation time with duration of 60s.:

- Scenario “1”: low-traffic (1-3 VoIP call), and
- Scenario “2”: high-traffic (4-8 VoIP calls)

Additionally, scenarios have been experimented varying the voice codecs which are previously described in Table 3.2. Note that we have integrated two other ITU-T codecs (G.726 and G.728) to *ns2voip++* framework.

Parameter	Value or Protocol
Propagation model	TwoRayGround
PHY/MAC layer	802.11b
Network layer	OLSR (RFC 3626)
Transport layer	RTP/UDP
Application layer	Ns2voip++
Voice codecs	ITU-T Codecs standard (Recall Table 3.2)
Call duration	60s
Simulation time	200s

Table 3.4: Simulation parameters

3.4.2. Results and analysis

This section shows and discusses our simulation results exhibiting the impacts of audio codec on QoS measured in terms of bandwidth, E2E-delay, packet loss and MOS.

3.4.2.1. Bandwidth

Figures 3.8 illustrate the bandwidth measured for different codecs which is increased in "*scenario 2*" because the network is more loaded. As can be observed G.711 gives the highest bandwidth versus other codecs, the opposite behaviour observed for G.723.1 and G.729A. The high bandwidth of G.711 codec can be explained by large voice payload used by this codec. Generally, a codec with higher bandwidth requirements provides better voice quality if the network capacity is enough to support. Hence, we have to make a trade-off between the VoIP codec and the available MANET resources spatially in term of the bandwidth.

3.4.2.2. E2E-delay

Figure 3.9 presents the average end-to-end network delay for different audio codecs. The results indicate that in a low-traffic scenario, all codecs does not exceed the threshold recommended by ITU-T (150 ms) [17]. This is due to network load (relatively low). While in high-traffic scenario, the lower bit-rate codecs (as G.723.1 and G.729A) present the best performance with respect to other codecs. These results are due to packet size. The larger packet size is the more time is required to process them. The relatively low payload (20 bytes) and transfer rate make G.723.1 and G.729A the ideal encoders. Otherwise, G.711 suffered higher delay than other coders for the reason that it has the larger packet size (160 bytes). In turn, we can observe that the E2E delay is increased when high bite rate codec is used.

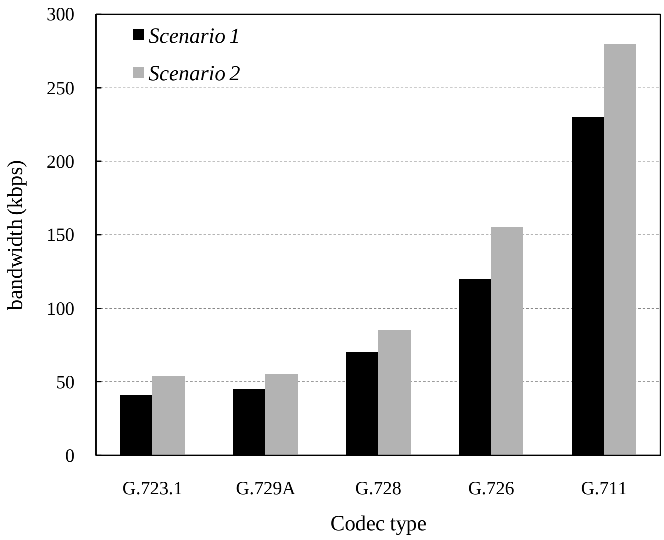


Figure 3.8: Bandwidth requirement for different audio codecs

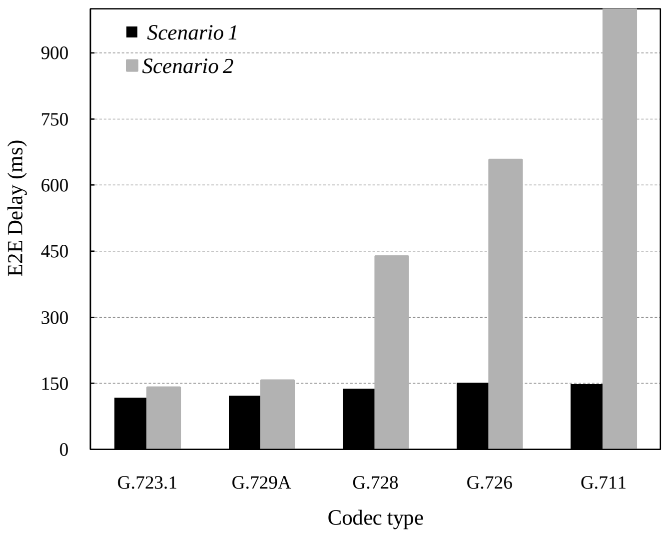


Figure 3.9: E2E-delay for different audio codecs

3.4.2.3. Packet loss

Examining the packet loss indicator for all codecs (Figure 3.10), we can check that it exceeds 10% which is the threshold for achievable voice communication. These losses are due to: firstly, the huge traffic introduced by the codecs in one second, while each nodes are only able to handle 50 packets in their queue. Secondly, bases on ns-2 simulation trace results, losses are due to route discovery process (*NRTE*) and routing loops (*LOOP*). While a node spends some time to locate a route to destination, the VoIP source continues to produce packets. When a route is not yet available, and the queue is full, packets on this one will be discarded. Codecs with low bit-rate is moderately decreasing packet loss in the static scenario because the traffic is slower (34 packet/sec) compared to other codecs. Generally, losses increase in high traffic scenario area because the number of connections increases.

3.4.2.4. MOS

The MOS is one of the most widely used QoS metric in VoIP applications, which help to computes a predictive estimation of the subjective voice quality. However, MOS is fundamentally affected by packet loss and delay. In figure 6, MOS is plotted for different codecs. The best MOS value is 2.8 for G.723.1 which seems quietly acceptable. In general, MOS values are not satisfied with the call quality, mostly due to high loss rate.

3.4.3. Synthesis

In summary, our simulation results have shown that the critical environment of the MANET negatively affect the performance of VoMAN. Since, the probability of link features tends higher because of fading radio propagation model. Based on investigating various scenarios, the most important observation is that performance is not satisfactory, especially in terms of packet loss. In fact, results have shown inability of codecs for a successful deal with VoIP applications requirement. In this situation, the major concern is to design and develop an efficient routing protocol for voice communication in such harsh environment which has very rapid changes. In this context, the next section investigates the impact of routing parameters on codec performance. This will be the first step of our solution design.

3.5. Understanding the impact of tuning OLSR routing parameters

Many properties of soft state signalling are not yet fully understood in the context of voice streaming over MANETs, especially their impacts on codec performance and the circumstances in which it might best be employed. This section focuses on the trade-off between audio codec performance and control overhead from the aspect of signalling. We assess the influence of tuning OLSR parameters on audio CODECs performance by caring out a series of simulations. Before starting the simulation study we provide a literature review of works having investigated the tuning OLSR.

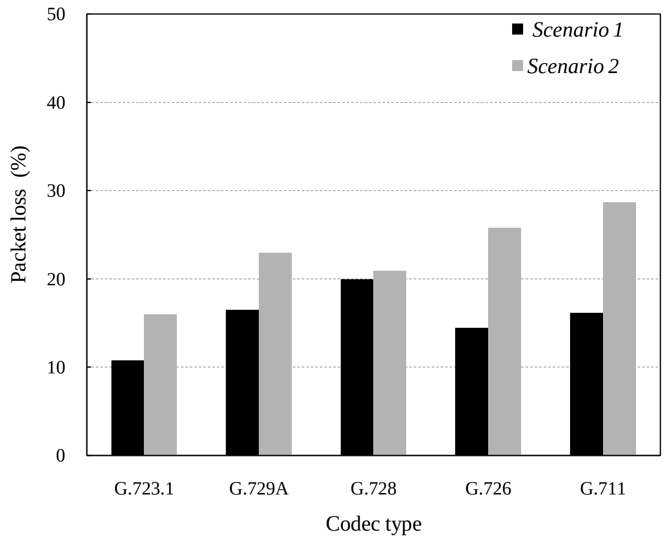


Figure 3.10: Packet loss for different audio codecs

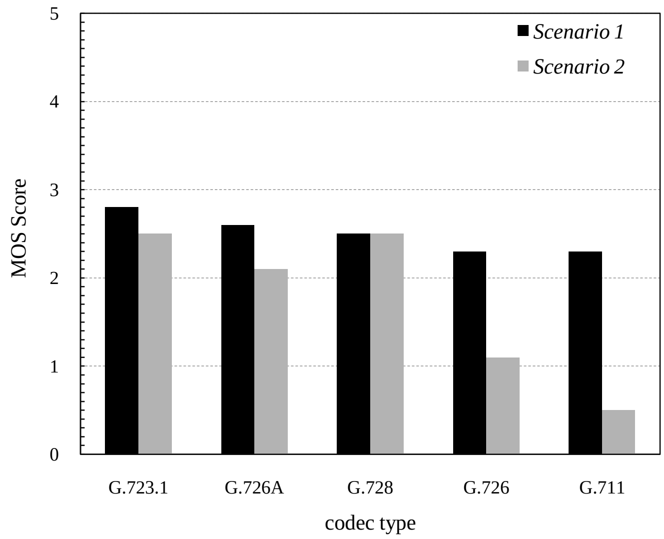


Figure 3.11: MOS score in MANET for different audio codecs

3.5.1. State of art

According to the results presented by several works, OLSR has a range of improvement by changing the configuration parameters. In [108], authors investigate the different impacts of tuning refresh interval timers on OLSR protocol performance; as a result, authors found that reducing refresh intervals could improve performance. Gomez et al. [109] analyses the impact of routing protocol parameter settings on OLSR routing performance. It studies the effects of tuning OLSR configuration (e.g. HELLO Interval and TC interval) on RCL (Route Change Latency), which is defined as the rerouting latency after a link failure. The study concludes that end-to-end connectivity can be enhanced using different parameter settings from the default ones.

Earlier work that address adaptation of OLSR for MANETs include [110], where Huang et al. proposes a self-tuning timer approach within a simple control system for MANET routing protocols with the aim of allowing dynamic, autonomic, re-calibration of routing update frequencies. A novel dynamic timer algorithm is presented to automatically tune routing performance by adapting timer intervals to network conditions. The results have shown that, compared to the default fixed timer approach, the proposed algorithm could effectively improve routing throughput without manual configuration. To the best of our knowledge, this is the first work that addresses the impact of tuning routing parameters on voice codecs performance.

3.5.2. Simulation scenario

Studies in wireless multi-hop networks highly depend on the scenario considered. This section proposes target scenarios for the study. The considered scenario is same as “scenario 2” in previous section. Table 3.5 summarizes some important features of the network used in our VoMAN simulations.

Parameter	Value or Protocol
Propagation model	TwoRayGround
PHY/MAC layer	DFC 802.11
Network layer	OLSR (RFC 3626)/Tuned OLSR
Transport layer	RTP/UDP
Application layer	Ns2voip++
Voice codecs	ITU-T Codecs standard (Recall Table 3.2)
Call duration	60s
Simulation time	200s

Table 3.5: Simulation parameters

At the network level, the creation of a multi-hop ad hoc network implies the need to choose an appropriate routing protocol to support reliable voice communications. OLSR protocol exhibits a series of features that make it well-suited for dynamic MANET. To compute the routing paths among the nodes, the network layer employs OLSR routing protocol with default parameter from RFC 3626 configuration. The maintenance of the internal state information held at nodes is directly related to the exchange of HELLO and TC messages. In this experiment, we tune two OLSR parameters: Refresh interval parameter (HELLO interval) and Willingness parameter. The simulation results are presented in the next section.

3.5.3. Refresh interval parameter impact

OLSR routing protocol are usually based on the link state (i.e., Dijkstra) algorithms. In this protocol, each node maintains routing information as topological repositories. Due to the frequent topology changes, the routing information has to be updated to guarantee the correctness of route selection. This requires exchanging messages between nodes, i.e. signalling.

In order to evaluate the impact of OLSR soft state signalling, in this section, we measure VoIP flow throughput, control traffic overhead (NRL), packet loss and E2E-delay for each codec, with respect to different HELLO message intervals (ranging from 0,5*s* to 3,5*s* by Δt=0,5*s*). The value of state holding timer intervals is adjusted correspondingly, and the TC message interval is set to a default value (5*s*). Note that, we have done some modification on OLSR implementation [107] to support decimal value for HELLO intervals. In the following, we present our simulation results:

Figure 3.12 plots average throughput that gives idea on bandwidth which needs to be occupied by each codec. We observe that increasing/decreasing neighbour detection intervals has not a significant impact on throughput for all codecs. However, G.711 has great throughput compared to other codecs, this is due to large packet size (160 bytes), which make a request for bandwidth. Concerning NRL Figure 3.13 shows that the overhead drops faster when increasing HELLO interval. However, the NRL is greater when using codec with high bit-rate. This confirms that the behaviour of OLSR is changed depending on the codec used.

From the point of view of E2E delay, as shown in Figure 3.14, HELLO interval doesn't affect delay for codec with low bit-rate. On the other side, (codec with high bit-rate) average delay changed significantly with increase/decrease of neighbour detection interval. We can see:

- When increasing HELLO interval, codecs with large sample size (as G.711) achieve good delay. This can be explained by lower control overhead releases the bandwidth which is requested by G.711.
- With the decrease of HELLO interval, the delay is lower for codecs having small sampling interval (as G.728 and G.726). These codecs generate voice frames in shorter time interval. So, high overhead is required to identify an appropriate path from the sender to the receiver.

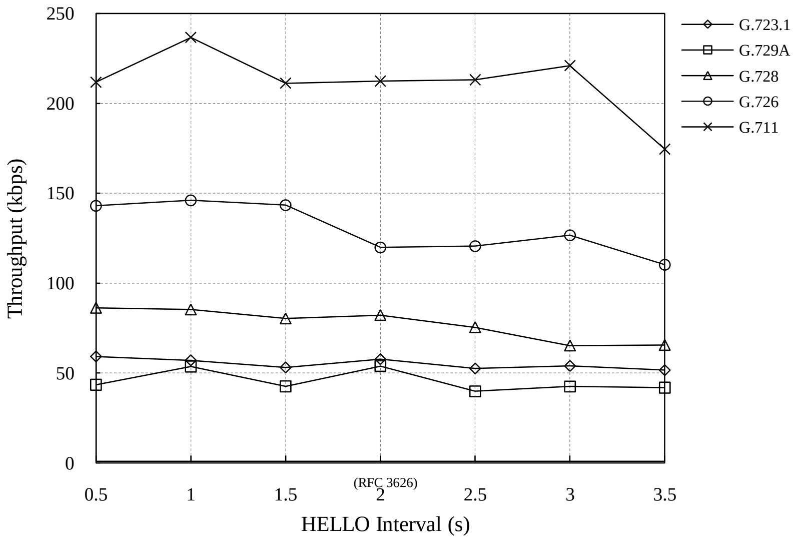


Figure 3.12: Throughput vs. HELLO Intervals

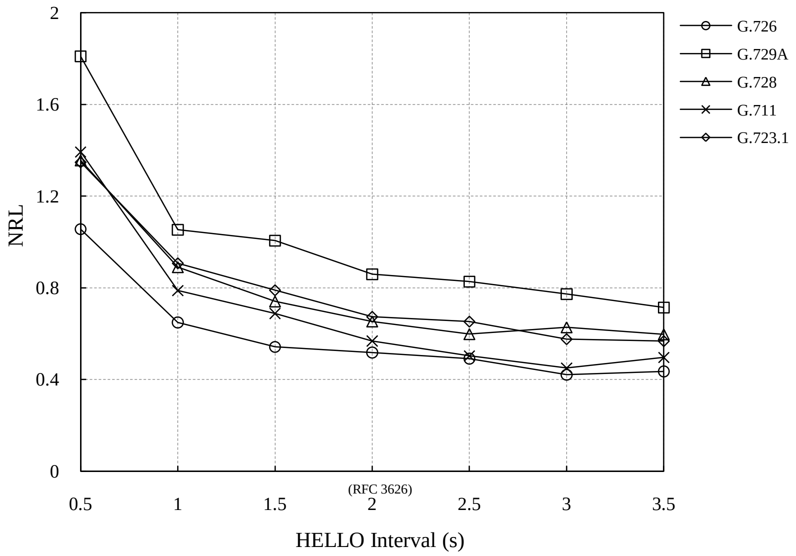


Figure 3.13: Normalized Routing Load vs. HELLO Intervals

In term of packet loss, values depicted in Figure 3.15 shows that reducing HELLO interval gives a small improvement. This is due to the high control overhead which leads to faster update of routing information. Therefore, it gives packets the ability to reach their destinations. Nevertheless, codecs must be more able to handle losses.

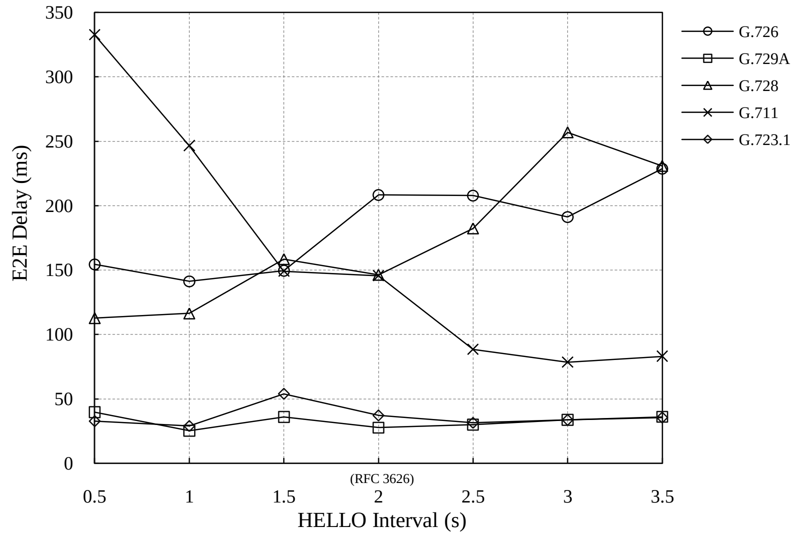


Figure 3. 14: E2E Delay vs. HELLO Intervals

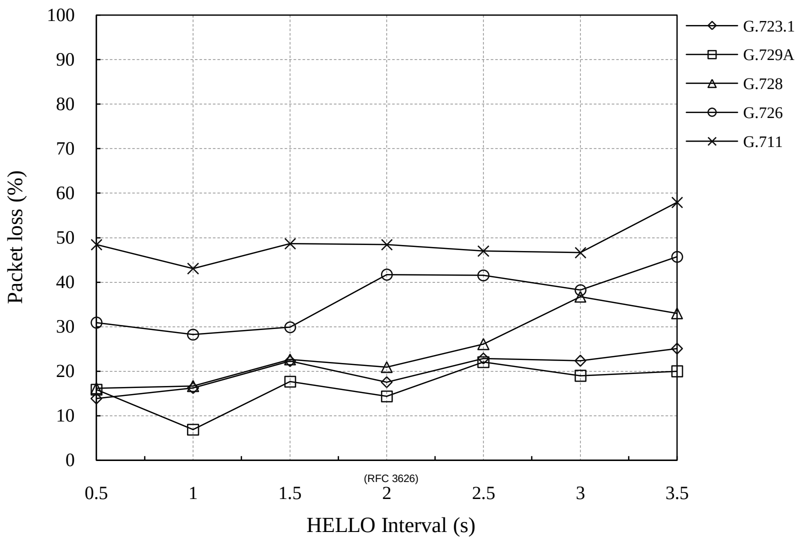


Figure 3.15: Packet loss vs. HELLO Intervals

Finally, according to the results obtained, adapting the HELLO intervals to codec setting could help in providing QoS for VoMAN. However, the performance depends on the accuracy of network measurement.

3.5.4. Willingness parameter impact

The Willingness is a parameter value announced by nodes participating in OLSR routed network to act as relays for OLSR control traffic for their neighbours.

The MPR selection algorithm selects first the nodes with the highest willingness. OLSR has eight values available for the Willingness: (from 0 "WILL_NEVER" to 7 "WILL_ALWAYS"). In this experiment, we tune the willingness parameter of nodes participating in a VoIP calls from 0 to 7. In order to show the impact of this parameter on codec performance, we measure E2E-delay, packet loss and NRL metrics.

Figure 3.16 and Figure 3.17 represent respectively the E2E-delay and packet loss with respect to varying willingness degree. From the above simulations it is seen that the E2E-delay and packet loss are increased when increasing willingness parameter. As known, high degree of willingness means that the node must perform other tasks on behalf of other nodes in the network, as: acting as MPR, forwarding data and TCs messages. So, increasing willingness in a VoIP node (node involving in a VoIP activity) adds an over-load in addition to VoIP load, which decreases considerably the bandwidth and causes delay and losses. Thus, greatly influences QoS. This gives an idea of what the gain can be when lowering the willingness degree of VoIP nodes.

The Figure 3.18 illustrates the NRL which is quasi-equals to 0.6 for all codecs. These results clarifies that the NRL is not affected by willingness parameter. From this analysis, we conclude that QoS in VoMAN depends on willingness parameter of VoIP nodes and not directly depends on codec used.

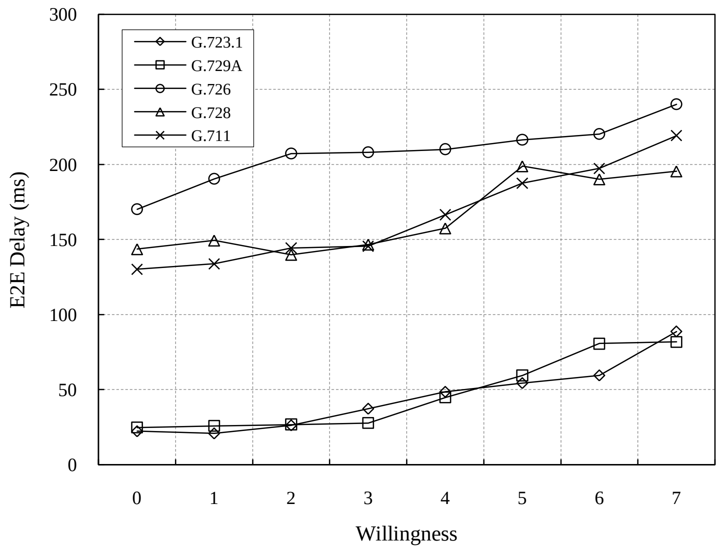


Figure 3. 16: E2D Delay vs. Willingness

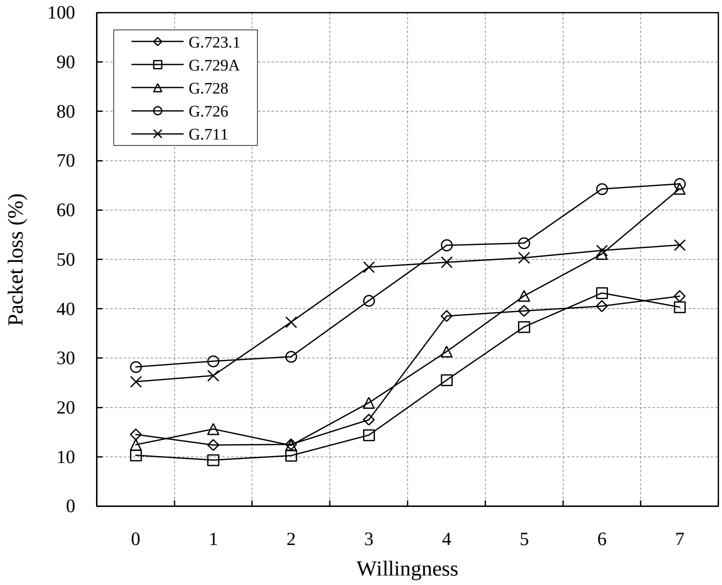


Figure 3.17: Packet loss vs. Willingness

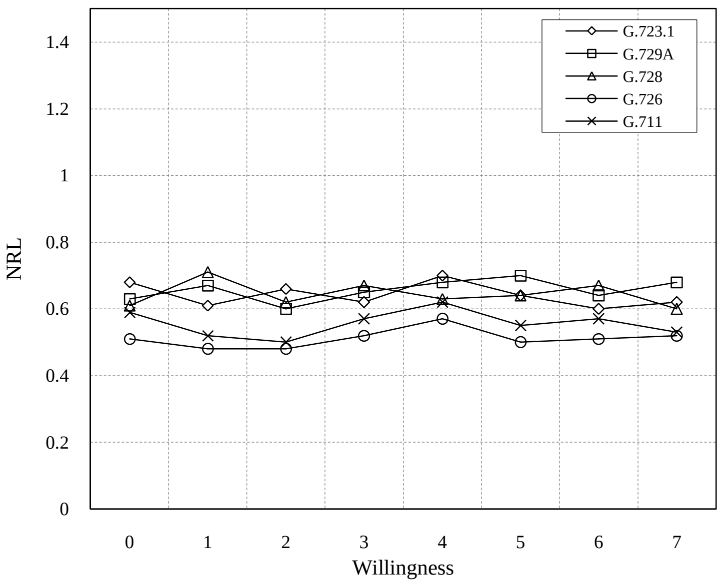


Figure 3.18: NRL vs. Willingness

3.5.5. Synthesis

In this study, we aim to gain a better understanding of tuning OLSR routing parameters impact on VoIP codecs. Through simulations, we are investigating two OLSR routing parameters:

Refresh interval timers: this parameter is used for periodic updates of link information and for subsequent topology maintenance. As results when decreasing the refresh interval this leads to increase routing control packet and thus the network bandwidth will be reduced. As known, VoIP codecs have different bandwidth requirements according to their voice payload and sample interval. So, configure hello interval to VoIP codec requirements may optimize the overall network bandwidth and increases VoMAN performance.

Willingness: is an important parameter to perform the task of routing. Simulation shows the dependences between this parameter and VoIP activities in the network. More specifically, for VoIP nodes where low value of willingness allows them to preserve resources for VoIP task. It may be possible to tune the willingness dynamically, during operation, by detecting node VoIP activity and configuring their willingness to WILL_LOW.

These results provide useful insights into how an adaptive routing protocol can be designed in the context of VoIP over mobile ad hoc networks. Such adaptive routing protocol require further investigation

3.6. Conclusion

Support for real-time multimedia session services, such as real-time interactive voice-over-IP applications, is highly desirable in a mobile ad hoc network (MANET). In this chapter, we have done comprehensive simulation studies which allow us to understand the behaviour of VoIP traffic in MANET. We began by confirming the effectiveness of proactive routing protocol for real-time application. Then we investigate different VoIP codecs and finally the impacts of tuning OLSR configuration were addressed, in order to make a trade-off between codec settings and routing parameters. Results have shown that VoIP performance is sensitive to routing parameters.

Routing adaptation mechanism may be an emerging solution for maximizing voice quality and provide suitable QoS for VoMAN. In the next chapter, OLSR routing modification will be performed, we will present a modified OSLR called OLSR-VA (OLSR-VoIP-Aware) and their benefits in improving voice quality over MANET by integrating adaptive routing parameters.

Chapter 4

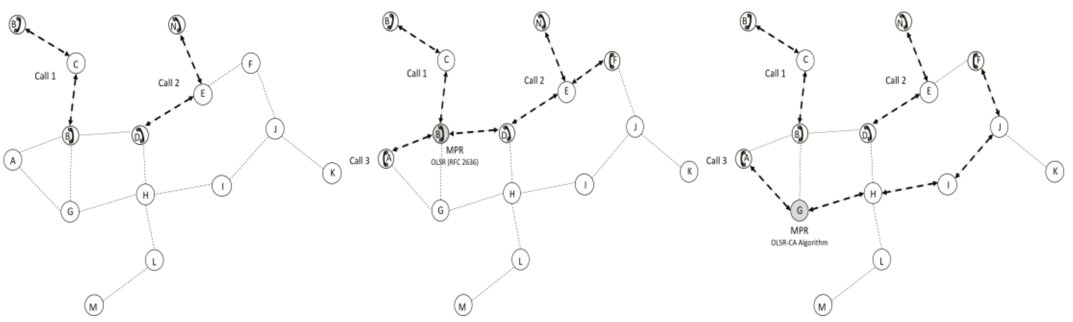
OLSR-VA : Making OLSR aware of VoIP

4.1. Introduction

The general purpose of our work is to provide MANETs the capability to stream voice traffic by adapting media transmission to changing network conditions. In this chapter, we propose a solution by providing self-adaptation mechanisms for OLSR routing protocol to adjust its parameters based on monitoring *VoIP Activity* (VA). First, we present our proposal approach, and then we describe in detail the protocol design. Finally, we validate the efficiency of our solution against the standard OLSR protocol, using simulation results for typical VoMAN scenarios. Our OLSR extension is called OLSR-VA (OLSR-VoIP Aware). In addition, we present a generic adaptive architecture that may be implemented in VoIP nodes, where performance gains can be accomplished through the exploitation of cross-layer designs and policy-based autonomic computing.

4.2. Proposal Approach

Our proposal for quality of service in VoIP over MANET has focused on adaptation between network layer and application layer by integrating self-adaptation mechanism in OLSR routing protocol. We propose a novel model for the OLSR protocol which exploits the signalling traffic to disseminate VoIP activity information in the network. Accordingly, each network element (node) will proactively has knowledge about VoIP activities happened in the MANET. The adaptation process is based on two actions:



(a) (b) (c)

Figure 4.1: VoMAN scenario

- **Monitoring** (feedback from application layer (VoIP activity and codec used) and network layer (routing)).
- **Adaptation** (self-tuning OLSR routing parameters (HELLO interval and Willingness)).

Consider a VoMAN scenario (Figure 4.1 (a)), with a number of VoIP call active (call 1 and 2). Consider now a new VoIP call (call 3) will be initiated between two users (nodes A and F). The audio codec is selected according to available resources (e.g. bandwidth). The OLSR routing protocol will find routes for VoIP traffic based on the shortest hop count paths first algorithm (e.g. path A-B-D-E-F), and will not care about VoIP load on this path (Figure 4.1 (b)). However, this lead to network parts with heavy VoIP load than others, presenting a high level of radio interference or a high level of congestion. The purpose of the extended OLSR protocol is to balance VoIP load in the network by routing VoIP packet over the less loaded paths even if requiring more hop count (e.g. choose path A-G-H-I-J-F, Figure 4.1 (c)). This process will reduce channel interference and congestion, and consequently the QoS will be improved.

This solution will make better bandwidth utilization. On the one hand, by adapting signalling overhead associated with OLSR protocol, and on the other hand, by transmitting voice packets via part of the network which has less VoIP activities. A key idea behind this solution is that the proposed solution can provide a part from a general QoS management architecture for VoIP over MANETs. A proposal with different policies combining the other adaptation techniques will be proposed in the last section of this chapter.

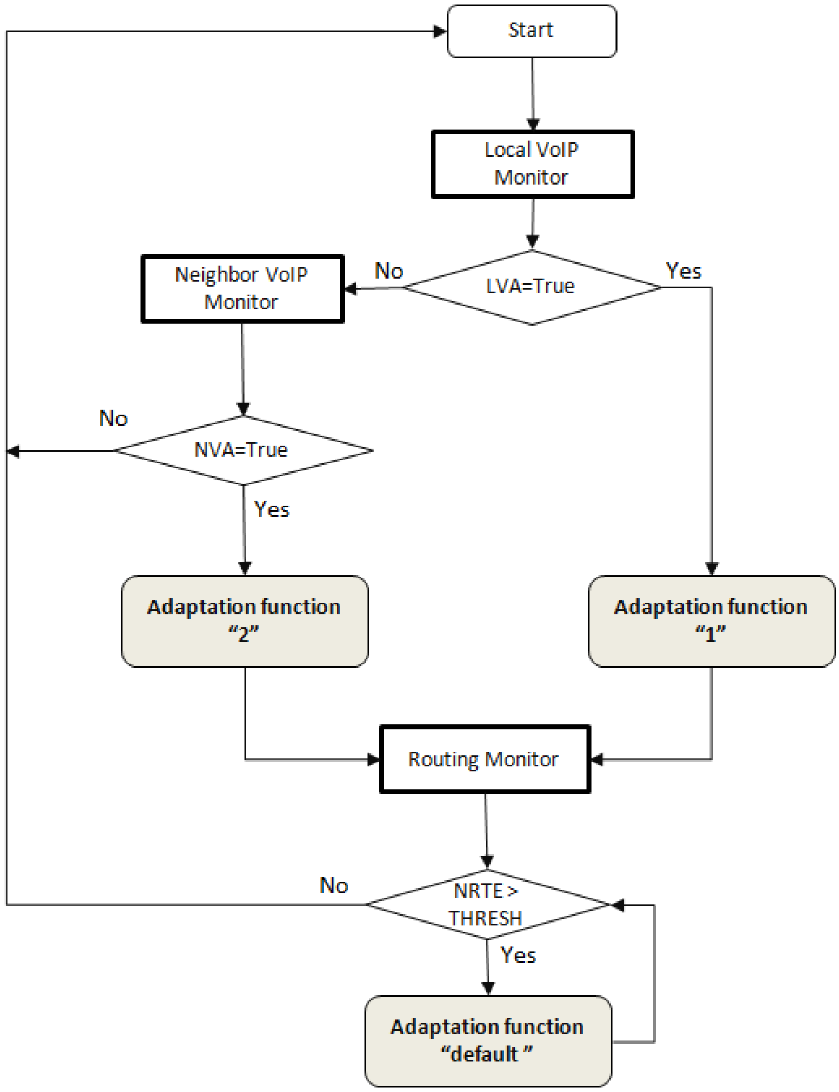


Figure 4.2: Algorithm flow chart

4.2.1. OLSR-VA Protocol design

Successful voice transmission over MANET needs the design of a routing protocol which takes in mind VoIP Activity (VA) and the audio codec requirements. This codec may be changed according to network condition. We propose an adaptive proactive routing algorithm that applies adaptation functions (varies the frequency of the neighbour detection messages and willingness in response to VoIP conditions) so as to achieve optimum voice quality with less control overhead. Essentially, the protocol's behaviour (i.e. parameters) is tuned according to VoIP activity and audio codec configurations. The basic structure of this proposal can be seen in a detailed algorithm flow chart in Figure 4.2.

In OLSR-VA, The monitoring phase is a constant triggers procedure, focusing particularly in three types of events: local VoIP activity detection (Local VoIP Monitor), neighbour VoIP activity detection (Neighbour VoIP Monitor) and routing performance (Routing Monitor). In the initial start-up phase, a node may have event triggers from the upper layer, informing a VoIP session establishment or a new received Hello message contains new information about VoIP session established in neighbourhood. Hence, the basic algorithm is based on three major "monitoring – adaptation" processes:

- *Monitoring Local VoIP Activity (LVA) - adaptation function "1":* in this process, the node monitors VoIP activity and codec changes from the upper layers, and then if appropriate applies adaptation function "1", or go to the next monitoring process.
- *Monitoring Neighbour VoIP Activity (NVA) - adaptation function "2":* in this process, the node monitors VoIP activity and codec changes from its neighbourhood, and then if appropriate applies adaptation function "2".
- *Monitoring routing performance - adaptation function "default":* this process occurs when adaptation functions "1" or "2" are applied. The node monitors the routing performance and then if appropriate applies the default adaptation function.

Each of these processes is analysed next in detail.

4.2.2. LVA Monitor - adaptation function "1":

Normally, OLSR sets up and maintains routes regardless of application layer communication demands. OLSR-VA monitors VoIP activity from upper layers using layer triggers mechanism. OLSR-VA is notified about a local VoIP session establishment and the audio codec used from a VoIP signalling protocol such as SIP [5]. If a local VoIP activity is detected, OLSR-VA has to adapt its routing parameters (HELLO interval and willingness) and disseminate VoIP information to neighbourhood. The resulting solution is presented in Algorithm 1.

HELLO interval is adapted according to the codec configuration (sample size and sample interval). When using codec with high sample size, MANET must guarantee bandwidth. However, routing control packets consume a good part of this one. Therefore, Increasing HELLO interval leads to some bandwidth preservation, obviously without affecting routing protocol performance. Codecs with small sampling interval generate voice frames in shorter time interval. So, high overhead is required to identify an appropriate path from the sender to the receiver.

Algorithm 1: OLSR-VA_LVA_adaptation_function (i, LVA_i)

Input: node i, Local VoIP Activity LVA_i.

Output: HELLO interval of i H_i

```
Begin
  if LVA_i = true then
    read codec parameters: sample_size, sample_interval
    if sample_size is high then
      Increase H_i
      else if sample_interval is high then
        decrease H_i
      end if
    end if
  end if
  W_i ← WILL_LOW
  return H_i
End
```

Willingness is a willing or interest of the node in the Ad Hoc network to give a contribution or commitment to the other nodes in order to send a data in the network. In OLSR-VA, each node, can declare an appropriate willingness. We decided to base the willingness selection on VA metrics. Willingness is set to WILL_LOW if the node has *LVA = true*. In this case, it will announce its inability to carry VoIP traffic on behalf of other nodes which makes provisions of their resource.

4.2.3. NVA Monitor - adaptation function "2":

The Neighbour VoIP Monitor is activated if there is no local VoIP session. The monitor simply process HELLO message received from neighbours, which contains information about VoIP activity in the neighbourhood. If the *NVA* monitor detects a *VA*, OLSR-VA applies the same adaptation functions as described in previous section, only the willingness is set to WILL_DEFAULT.

The modification of willingness leads to heuristic routing mechanism using a new algorithm for MPR computation. In OLSR (RFC 2636), each node selects their MPR permitting to compute the routes based on the knowledge of state of the network. The original heuristic for MPRs selection constructs the MPR-set that enables a node to reach any node in the symmetrical strict 2-hop neighbourhood through relaying by one MPR node with willingness different from WILL_NEVER. However, In OLSR-VA, heuristic routing allows a measure of route optimization based on recent knowledge of the state of the network and also VoIP activities occurring on it (*VA* metric). Therefore, the MPR must be selected in such a way that they will ensure voice transmission over links presenting the less VoIP activity. The following Algorithm 2 specifies a proposed heuristic for selection of MPRs.

Algorithm 2: OLSR-VA_MPR_selection (i, N_i^1, N_i^2)

Input: node i, 1-hop neighbour N_i^1, 2-hop neighbour N_i^2, 1-hop willingness neighbour W_i^1.
Output: MPR set of i M_i.

Begin
  if nodes in N_i^1 are the only nodes to provide reachability to a node in N_i^2 **then**
    Add nodes to M_i
  end if
  if nodes in N_i^2 are not covered by a node in M_i **then**
    Remove nodes from N_i^2
  end if
  while $N_i^2 \neq \emptyset$
    foreach node in N_i^1,
      Calculate R (the number of nodes reached in N_i^2)
      if R is highest or W is highest **then**
        Add node to M_i
      end if
      if a node in N_i^2 is reachable by more than node in N_i^1 **then**
        Add to M_i node having highest W
      end if
    end each
    Remove from N_i^2 the nodes that are now covred by a node in M_i
  End while
End

4.2.4. Routing Monitor - adaptation function "default":

Routing Monitor is activated after one of the adaptation processes discussed above is applied. As previously described, OLSR-VA is an adaptive routing protocol which adapts their parameters with respect to events (*LVA* and *NVA*). If this adaptation leads to routing performance degradation, OLSR-VA must tune their parameters to defaults ones (that of RFC 2636 standard).

NRTE metric is used to evaluate routing performance. This metric represents the number of packets dropped because of no available routes, which means that the routing protocol is unable to forward packets to their destinations. We calculate the average of the number of packet dropped during 5 seconds. If the average is greater than 100 packets the default adaptation function (OLSR RFC-2636 default parameters) will be applied. The *LVA* and *NVA* Monitors can't be activated until achieving an average less than 100 packets.

Algorithm 3: OLSR-VA_Default_adaptation_function (i, $NRTE_i$)

Input: node i, number of dropping packet cause of no available route reason $NRTE_i$.

Output: default parameters values

Begin
  if LVA_i = true or NVA_i = true **then**
    for t = $Current\_time$ **to** 5
      Calculate $AVG(NRTE_i)$
    End for
    While $AVG(NRTE_i) > 100$ **do**
      $H_i \leftarrow$ defout_values
      $W_i \leftarrow$ defout_values
    End while
  End if
End

4.2.5. Dissemination of VoIP Information

As explained in Section 2.3.6.2, to establish and maintain the OLSR repositories Information, a number of different OLSR messages are defined and exchanged periodically by the nodes participating in the network. Together they form the OLSR control traffic. OLSR exchanges periodic HELLO messages and collects 2-hop neighbourhood and MPR information to be able to construct the routes. This mechanism can be easily extended to carry the VoIP information as well. The format of the HELLO message (RFC 3626) illustrated in Figure 4.3 contains two sections:

- **Local information section**
  - Reserved (2 bytes): unused fields and filled with zeros.
  - Htime (1 byte): holds HELLO emission interval (HELLO INTERVAL), the time until the next HELLO message transmission.
  - Willingness (1 byte): defines the willingness of a node to carry or forward traffic on behalf of other nodes.
- **Link information section**
  - Link-code (1 byte): contains both info about the link to the neighbour and the type of the neighbour.

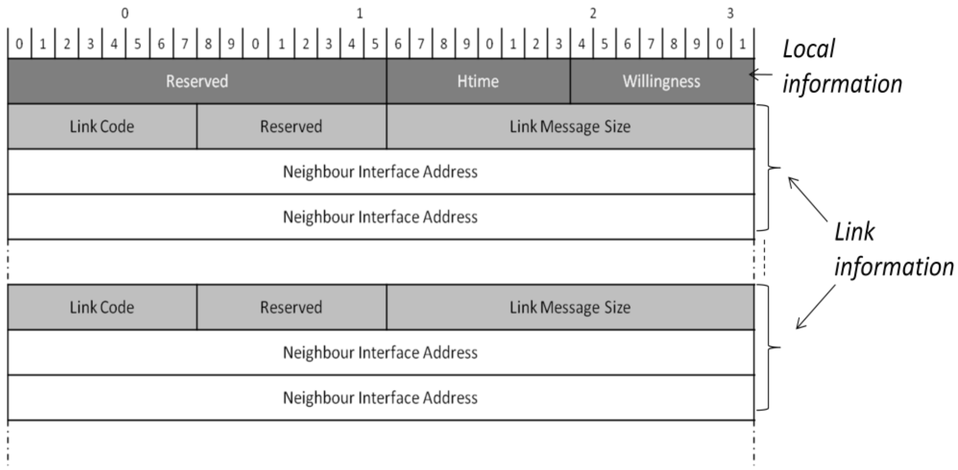


Figure 4.3: OLSR HELLO message

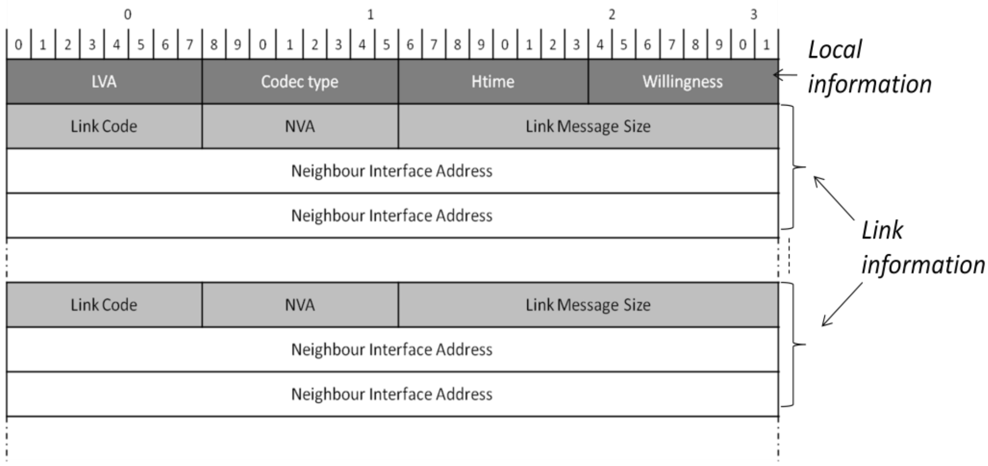


Figure 4.4: OLSR-VA HELLO message

- Reserved (1 byte): unused fields and filled with zeros.
- Link Message Size (2 bytes): specifies the message length between two consecutive Link Code
- Neighbour Interface Address (4 bytes): specifies the address of the neighbour node's associated interface.

To implement OLSR-VA core functionality new HELLO message type has to be supported. Thank to Reserved fields, Original Hello message structure can be easily extended to supply the necessary information of VoIP activities for the originating node itself and its listed 1-hop neighbours. The proposed modified message structure is shown in Figure 4.4.

We propose using the first half of the reserved field within the local information section for signalling local VoIP activity (*LVA*) (1 byte) and the second half for voice codec used (1 byte). Hence, in addition to the original local information (such as willingness and Htime) node also includes its own *VA* information in the HELLO message. The Reserved field in the link information section is used for signalling 1-hop neighbours VoIP activity (*NVA*) (1 byte); i.e. upon receiving a HELLO message from its neighbour, a node reads the neighbour LVA value, and includes its neighbours *VA* information in the HELLO messages.

As result, through the exchange of this new HELLO message structure, there is no extra overhead introduced as the unused parts of HELLO messages are utilized for the dissemination of the VoIP activity information.

4.3. Performance evaluation: Urban VANET scenario case study

What advantage might adaptation offer? In order to answer this question, in this section, we study the performance of OLSR-VA considering concrete scenario of an urban VANET environment. We chose this scenario because it presents a critical situation where voice transmission is needed and the network infrastructure is unavailable. Examples of these situations concern dynamic communication in emergency search, disaster rescue operations, and a battlefield (Recall Figure 1.1).

The proposed solution is evaluated using simulations on the ns-2 network simulator [99] version 2.34. Considering the VANET scenario, we must use specific mobility and propagation models which reflect a real urban environment. We compare the performance of our scheme with that of the original protocol, to show the improvements in the performance. The performance metrics include tow metrics (quantitative and qualitative metrics) will be defined later. Firstly, we provide an overview of technical aspects of VANETs in terms of architecture, routing and MAC protocol, and mobility model.

4.3.1. Vehicular ad hoc networks: Overview

Vehicular Ad hoc NETworks (VANETs) are specific class of MANETs that are used to connect vehicles in urban environments or highways to promote safe and comfortable driving. However, such networks introduce several constraints like the high mobility of the nodes, frequently changing topology, and unpredictable delay [111]. These characteristics distinguish them from other mobile ad hoc networks and make the transmission of multimedia traffic over such networks a challenging task.

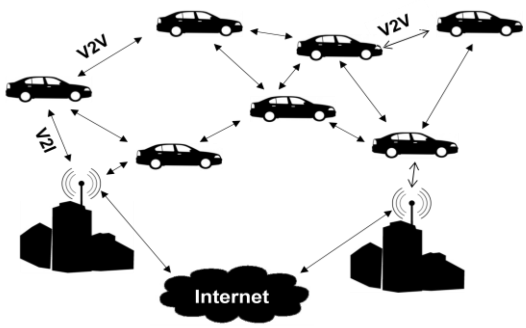


Figure 4.5: VANETs system architecture

From the vehicular communication perspective, VANETs system architectures (Figure 4.5) can be categorized into vehicle to infrastructure communication (V2I) systems, and inter-vehicle communication (V2V) systems.

From network layer perspective, routing protocols find the appropriate route in a wireless multi-hop environment. They need to take in consideration the high speed mobility and the unpredictability of VANETs. MANET routing protocols need to be redesigned in order to be applied to VANETs needs. Recent studies show that proactive routing protocols such as OLSR generally outperform the reactive ones in terms of network goodput and end-to-end delay (that is an important feature for VANET application) [112].

From data link perspective, various MAC protocols have also been proposed in order to efficiently share the medium in VANETs networks. Currently, DSRC (Dedicated Short-Range Communication) implements the IEEE 802.11p wireless standard which gives physical and MAC level specifications for VANETs [113]. 802.11p enhances wireless access functionality that will permit applications for rapidly changing vehicular network environments. At the physical layer, IEEE-802.11p operates at 5.9GHz band (U.S) and 5.8GHz band (Japan and Europe) with 75MHz bandwidth divided in seven channels each with 10 MHz frequency band. The center channel is a control channel and the rest are the service channels.

From mobility model perspective, classical mobility models for MANETs (like Random Way Point model) may not be directly applicable to VANETs, because vehicular environment presents different requirements, such as constrained road topology, multi-path fading and roadside obstacles, varying vehicular speed and mobility, traffic lights, traffic congestion, drivers behaviour, etc [113]. However, to achieve good results from VANET simulations, realistic mobility model (that is as an actual vehicular environment) need to be generated. Currently, different road traffic generators have been used in order to generate the realistic simulation mobility models where vehicles move following the real traffic rules. Martinez et al. [114]

provides a comparative study of various publicly available VANET simulators and mobility generators that are currently in use by the research community.

These entire factors were being considered in our simulation scenario. The following subsection proposes target scenarios for the study.

4.3.2. Simulation scenario description

In this study, a road traffic generator was used combined with ns-2, aiming at a significant level of simulation accuracy. The traffic simulator is needed to generate realistic vehicular mobility traces, used as an input for the network simulator. In order to increase the level of realism in VANET simulations, the simulations use a microscopic vehicular traffic generator based on the car-following and lane-changing models proposed by Gipps et al. [115], which belong to the class of collision avoidance vehicular mobility models. This generator is used in conjunction with ns-2 and digital road maps from the XML data of Open Street Map [116].

To generate trace files reflecting vehicles movements, we consider a typical urban scenarios (portion of the area of Tangier city, in Morocco), presented in Figure 4.6. The scenario focuses on the unicast transmission of voice signals between vehicles moving at rate of 0-50 km/h, with an average inter-vehicle distance of 5 to 20 meters in increments of 7 meters. In order to analyze how various conditions affect the quality of the voice, two scenarios are considered (details are summarized in Table 4.1). Each scenario defines a network area sizes, which is simulated with varying conditions: size area, VoIP traffic and densities.

The IEEE 802.11p standard is used in this experiment, and has been developed in recent ns-2 versions. Thus, the configuration is completed by introducing two new native modules: *Mac802_11Ext* and *WirelessPhyExt* that have been developed in [117]. The extension s are based on *Mac802_11* and *WirelessPhy*, but did a major modification to the original code. Propagation signal modelling is a fundamental issue on wireless simulation studies. Recent research has shown that a fading radio propagation model, such as the *Nakagami* model, is best for simulation of a VANET environment [118]. As a result, simulations carried out in this study, have been configured to use this propagation model.

In application level, *ns2voip++* modules were used to generate VoIP traffic. The VoIP source is configured to draw the duration of the talk-spurt and silence periods from *Weibull* distribution. A number of different codecs are considered. To this end, we consider that end users support multiple VoIP codecs. Table 4.2 summarizes the important features of the network used in our simulations.

	Scenario "1"			Scenario "2"		
Area size	500m²			1000m²		
Vehicles	10	20	30	30	40	60
VoIP calls	2	5	10	10	15	20

Table 4.1: VANET simulation scenarios

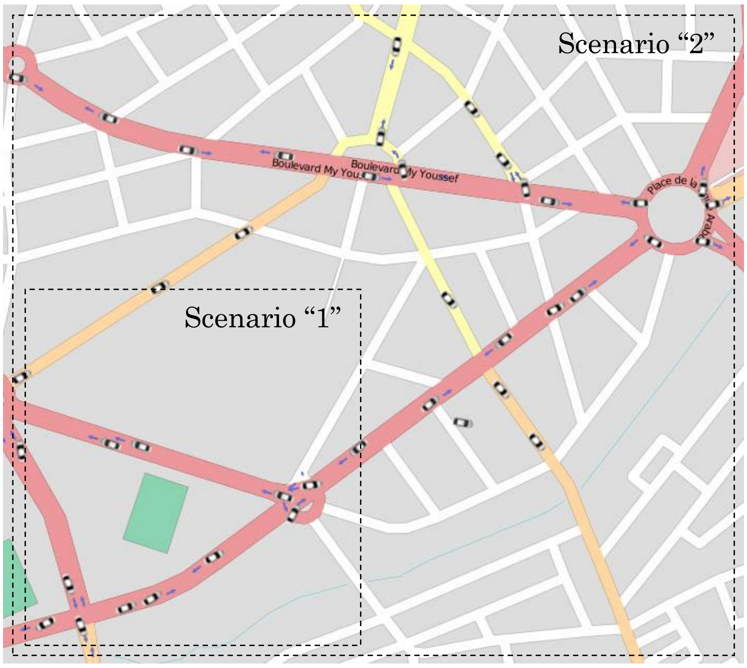


Figure 4.6: Tangier urban areas taken into account in our experiments

4.3.3. Evaluation metrics

We aim to evaluate the performance of OLSR-VA gathering both quantitative and qualitative metrics of VoIP quality:

- **Quantitative metric:**

This metric quantify the quality degradation of VoIP packet (QDVP), and then gives the percentage of lost and late packets measured at the destination. As a result, $0 \leq QDVP \leq 1$, while the smaller the QDVP the better. Based on [119] this metric is defined as:

$$QDVP = \frac{P_{lost} + P_{late}}{P_{total}}$$

Where: P_{lost} is the number of packets lost, P_{late} is the number of packets arriving to their destination after 150ms, P_{total} is the total number of packets sent.

NB. QDVP does not take delay variation into account explicitly; the receiver can employ a play-out buffer to smooth out variations in packet arrival.

- **Qualitative metric**

QDVP metric reflect the total performance measurement at network level, while the measurement at user level (perceived voice quality) is not clear. To this end, we use the subjective metric, Mean Opinion Score (MOS), which is a numerical indication of the listening quality of the received audio stream. MOS calculation method is presented in detail in Section 2.2.4.

Simulation parameter	**Value or Protocol**
Simulator	Ns-2.34
Simulation hardware	Intel C2 duo CPU E4400 at 2 GHz, 4GB RAM
Simulation time	500s
Simulation warm-up time	100s
Transmission Range	250 m
Fading model	Nakagami
Topology model	Urban VANET
Call duration	60s
Protocol parameter	**Value**
Routing protocol	OLSR (RFC 3626)/ OLSR-VA
PHY/MAC protocol	802.11p
Transport protocol	RTP/UDP
Buffer size	100 packets, drop-tail queuing policy
Application layer	Ns2voip++
Voice codecs	ITU-T Codecs standard (Recall Table 1)
Talkspurt /silence periods	Weibull distribution
Default OLSR- VA parameter	**Value**
HELLO Interval	3s
TC Interval	5s
Willingness	3
LVA	False
NVA	False
NRTE_TRESH	100

Table 4.2: Simulation and protocol parameters

In the following section, we show and discuss our simulation results investigating the impacts of the introduced routing mechanisms.

4.3.4. Results & Analysis

The overall system performance was tested using two sets of measures. In the first one, the network does not implement any adaptive algorithm and OLSR routing protocol is used. In the other set of simulations, each mobile node provides OLSR-VA functionalities. For each simulation scenario we perform 10 runs with varying random simulation seeds (initial node position in simulation area). We measure QDVP and MOS metrics in function of simulation time (500s). The VoIP calls are started after the 100s of network simulation warm-up. In order to show the impact of our adaptive algorithm we switch the VoIP codecs randomly after 250s of the simulation time.

We first present simulation results for scenario “1” and then for scenario “2” (scenarios described in Subsection 4.3.2). Figure 4.7 plots the QDVP and MOS as a function of the simulation time for both OLSR and OLSR-VA. Measures are taken every 10 seconds and each value represents the average of 10 simulations repetitive results. As it can be observed, before applying codecs switching (i.e. before 250 seconds of simulation time), both protocols have the same behaviour whether for QDVP or MOS, and the maximum MOS obtained is about 3,4. When the variation phase of codecs starts (after 250s), results clearly shows that OLSR-VA deal perfectly with the changing codecs. These can be explained by the adaptive behaviour of OLSR-VA with regard to codec used in each period of the simulation time. While changing codec affect the OLSR performance. As it can be seen, the MOS goes down to 3 which is unacceptable value for our application interest (voice communication in emergency response).

DVPQ and MOS almost appear as a mirror image, but they are not always presented in a mutually proportional manner. For example, in 340s of the simulation time, the QDVP value of OLSR-VA is 0,37 and MOS value is 3,5, while in 370s MOS obtained is 3,8 for the same QDVP value (0,37). This can be explained by the calculation of MOS which is based on metrics used in the calculation of QDVP (delay and packet loss) in addition to other parameters related to codec (such as codec impairment).

In Scenario “2”, the analysis focus on the results considering a large urban scenarios sizes with a significant increase in density of vehicle and VoIP calls. Figure 4.8 shows simulation results exhibiting QoS measured in terms of DVQP and MOS metrics. In general, the behaviour of the OLSR and the adaptive scheme OLSR-VA shows degradation in quality. This can be explained by the critical environment of the scenario which could negatively affect the network performance. Since, the probability of link features tends higher because of fading radio propagation model.

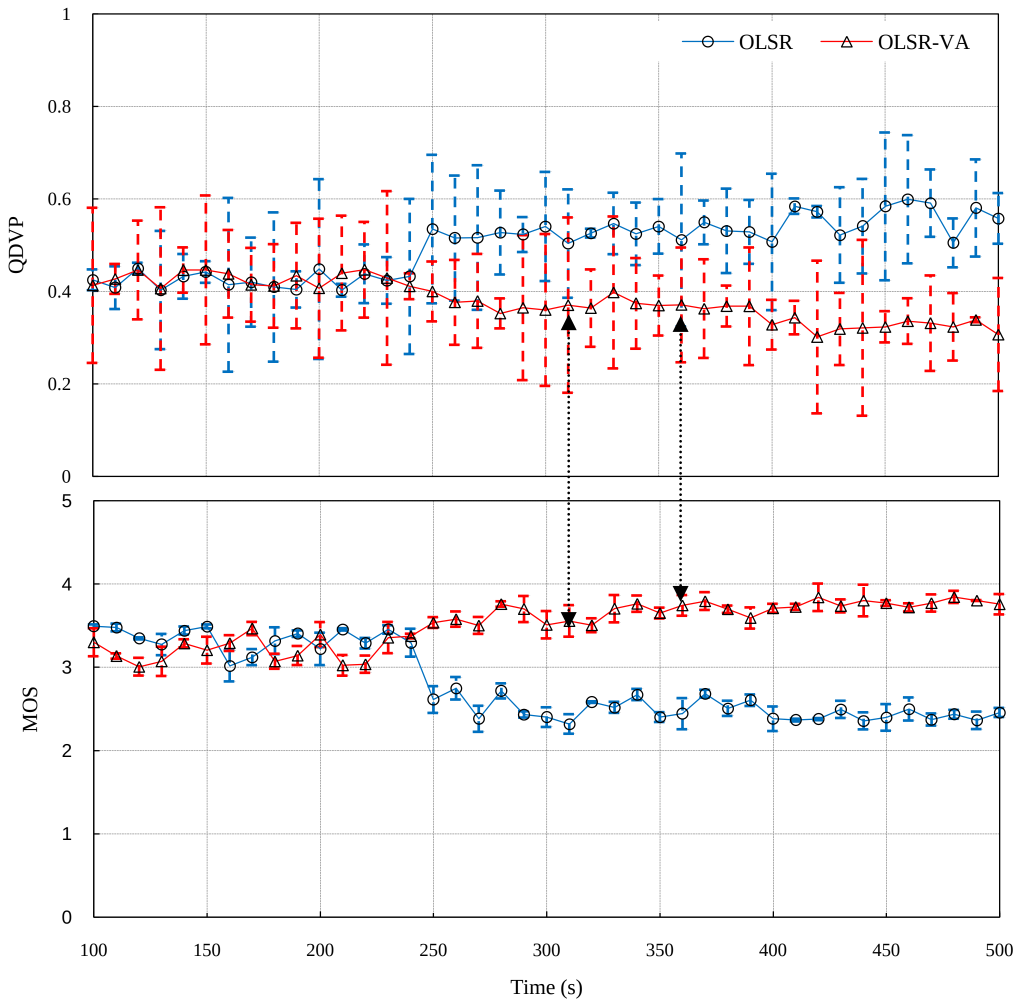


Figure 4.7: QDVP and MOS as a function of simulation time in scenario “1”

Moreover, in this scenario, longer paths are expected which result in longer delay due to more queuing at intermediate hop. In turn, OLSR-VA keeps its distinctiveness in relation to OLSR, where it was able to remain the desired minimum value of the MOS. However, in this situation, the major concern is to design and develop an efficient solution for voice communication in such harsh environment.

In summary, simulation results have shown that the proposed scheme gives better performance compared to traditional approaches following the layered architecture by selecting paths with high bit-rate links while also avoiding areas of MAC congestion. Additionally, based on investigating various scenarios, our solution OLSR-VA performs better than OLSR to deal with codecs change, and provide acceptable voice quality.

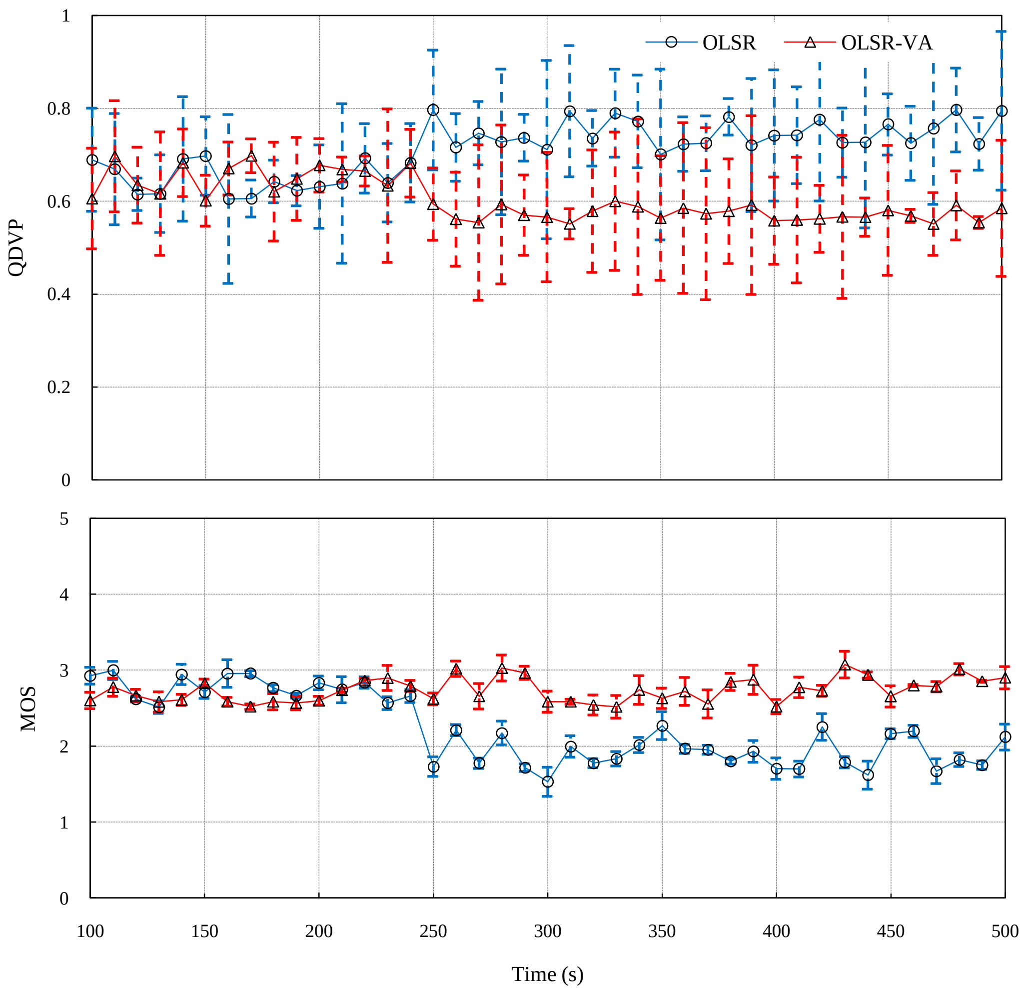


Figure 4.8: QDVP and MOS as a function of simulation time in scenario "2"

4.4. Call capacity evaluation: Healthcare scenario case study

Communication technology plays a central role to improve healthcare quality. It provides the foundation needed to allow doctors, nurses and support staff to get closer to their patients to accurately record data in real time, provide care and deliver medication and/ or treatments. VoIP over MANET is another kind of application that may either be used in emergency rescue missions where participants (patients as well as doctors) in a healthcare system may form a healthcare VoIP network that is ad hoc in nature. In this section, a healthcare scenario is addressed. We evaluate our solution for an environment where normal network connectivity may not be available hence voice over e-health networking becomes necessary.

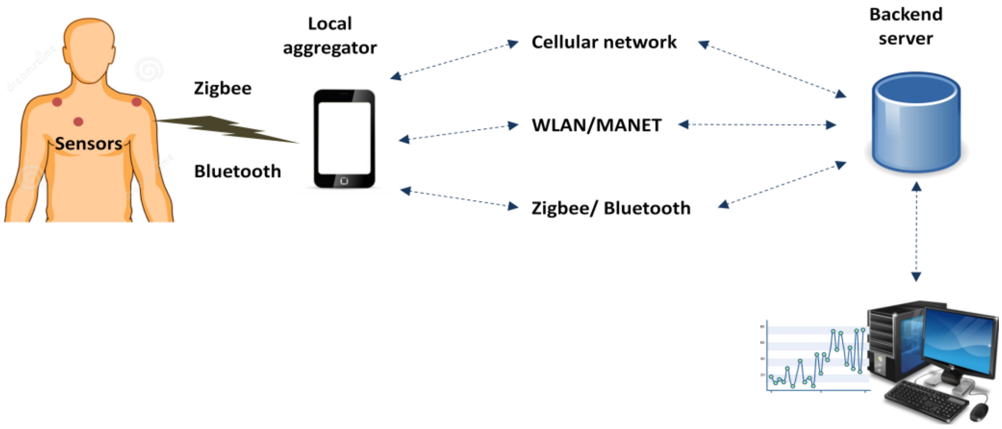


Figure 4.9: Wireless healthcare system architecture

4.4.1. Healthcare and wireless technologies

4.4.1.1. Overview

The healthcare industry incorporates several sectors that are dedicated to providing healthcare services and products. As a basic framework for defining the sector, healthcare can be categorised as generally consisting of hospital activities, medical and dental practice activities, and other human health activities. Using communication technologies, a wireless healthcare system provides tools to screen, monitor and manage general consumer health, high-risk and chronically ill patients, and the wellness and fitness communities.

The Body Sensor Networks (BSN) are well known wireless healthcare applications which could allow inexpensive and continuous health monitoring with real-time updates of medical records. As shown in Figure 4.9, Wireless healthcare system architecture is consisting of a set of wireless sensors, a wireless local aggregator (LA) for each user and a backend server (BE).The main mode of operation is to gather the data from the sensors at the local aggregator and transmit it to the backend for further processing. This system provides continuous monitoring and analysis of physiological state of the patient.

Wireless technologies provide healthcare organisations with mobility capabilities to better achieve their goal of delivering high-quality medical care. These technologies largely rely on infrastructure based wireless networks. However, in many cases, such as field hospitals area (large mobile medical unit that temporarily takes care of casualties on-site), have low or non-existent coverage from infrastructure based networks. Hence, mobile ad hoc network is a natural solution that can be used to support this kind of nomad activities and meet all of the data and voice requirements for healthcare applications. With a large mobile population of doctors, nurses, physician's assistants and other caregivers, MANETs bring the ability to access the latest patient charts, medical records and clinical decision support data at all times, anywhere in the healthcare organisation. And as caregivers travel among different facilities, wireless allows for easy connectivity at each site.

4.4.1.2. VoIP over healthcare ad hoc network

As is the case for data exchange, voice communications are needed in healthcare applications. Voice over MANETs enhances mobility further, enabling doctors and other staff members to seamlessly move from one place to another during emergency situations, all while talking on a mobile VoIP handset or through a voice enabled badge. VoMAN solution provides many advantages including:

- Never miss a call, even in disasters or outages – In the event of a power or circuit failure, VoMAN pre-set disaster call routing feature ensures that your incoming calls from patients or staff still reach users at all their designated locations.
- With VoMAN phone systems for field hospital, healthcare worker can easily transfer calls between exam rooms, set up a paging group to communicate across the area.
- Often, the call can't simply be ignored. For example, if a patient has a heart attack or other medical emergency, the nurse is forced to respond.
- VoMAN can help to reduce the network deployment cost, hence call tariffs significantly cost lower.

Since field hospitals are so portable, having good interoperability between hospitals in neighbouring districts and the police, ambulances, and fire-fighters, is a necessity. Having a fast deployed inter VoIP communication system is also an important element to a field hospital, since the doctors and medical staff working there, are normally working under dire and life- threatening conditions.

In next section, we study an ad hoc healthcare network scenario. We evaluate our solution for a field hospital environment where normal network connectivity may not be available hence voice over e-health ad hoc networking becomes necessary.

Figure 4.10: Healthcare scenario in a field hospital

4.4.2. Simulation scenario

As illustrated in Figure 4.10, VoMAN solution feature is designed to convert the field hospital into a single, low-cost, unified communications solution that handles voice calls. The capability lets healthcare workers make and receive phone calls when connected to their internally managed ad hoc network. However, in this situation, healthcare workers need to act and communicate with each other in field hospital area that significantly differs from the usual scenarios.

In this scenario, we consider 50 participants (doctors, nurses, patients, etc.) in a field hospital area of 500m^2. Each participant is equipped with a mobile VoIP device (handset or a voice enabled badge) and forms a node of the network. The simulation is based on the same parameters used in previous section except for the mobility model. Since obstacles placed in the network area pose limitations to nodes mobility and signal propagation, in our scenario we must consider a mobility model which simulates user movement of real-healthcare applications in a field hospital area. In doing so, we have used the Random Trip Model (RTM) which contains of many particular mobility models, including the widely known Random Waypoint and Random Walk. The typical scenario consists of slow moving objects (caregivers staff) that searches field randomly on foot (0-5 m/s).

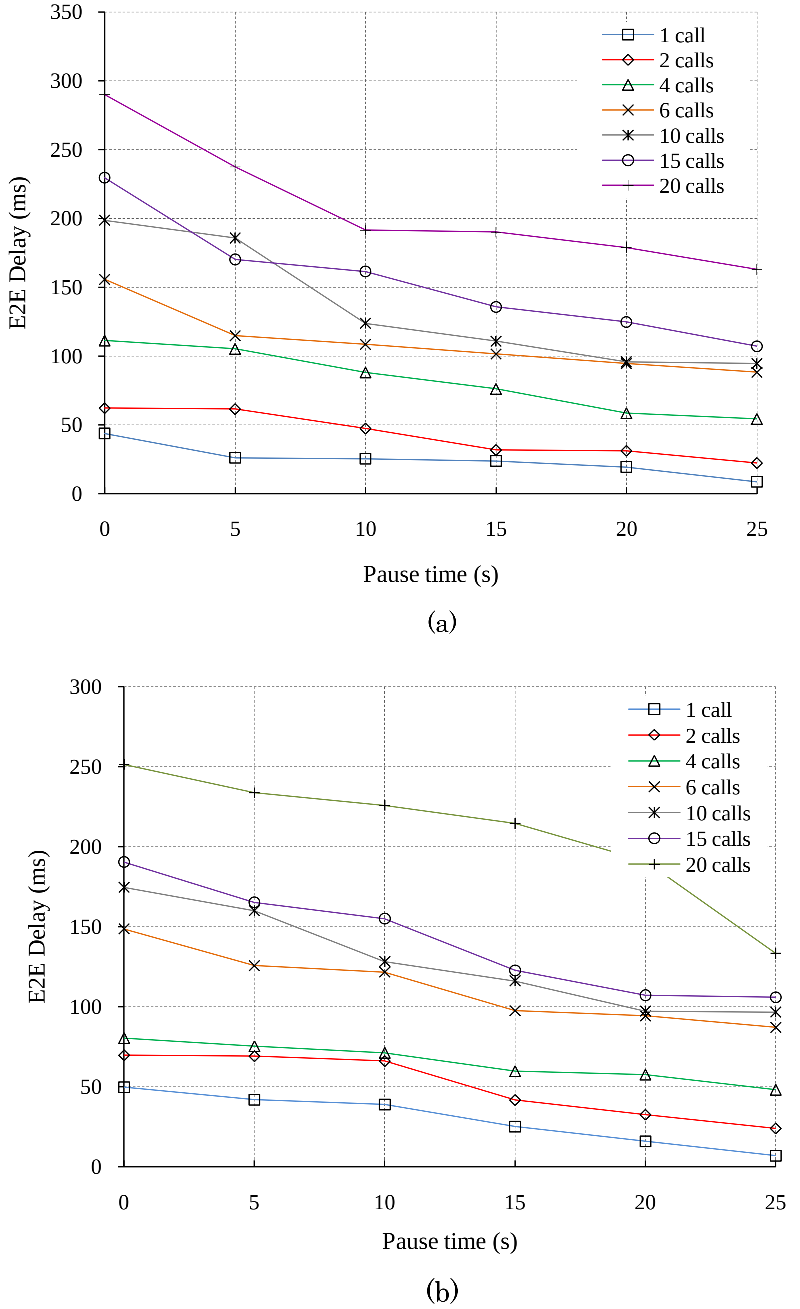


Figure 4.11: E2E Delay vs. Pause time (a) for OLSR (b) for OLSR-VA

Already, the capacity of voice calls is reduced by DCF mechanism [21]. Additionally, in networks with high VoIP calls, a substantial percentage of link capacity is wasted due to control overhead. Alternatively, we have to investigate quality enhancement which does adaptation offer under a given load conditions. In this scenario, our target is to investigate whether the adaptive routing protocol solution (OLSR-VA) has any benefit over the non-adaptive one in term of call capacity.

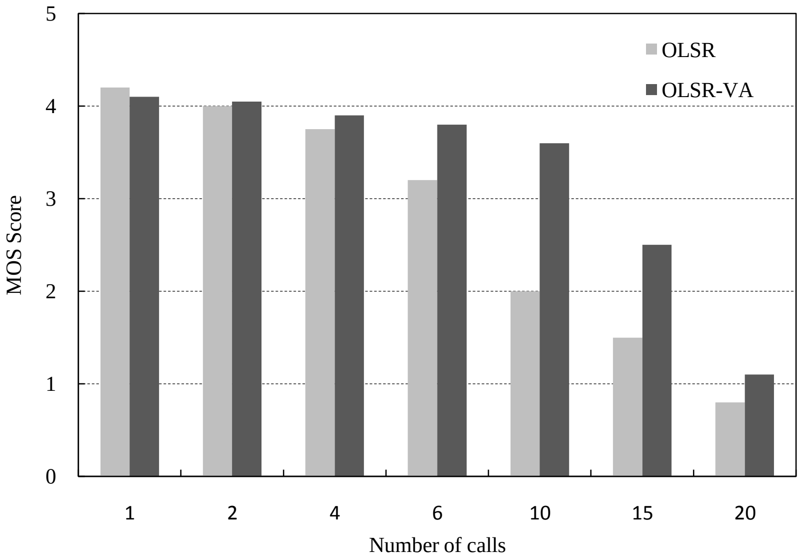


Figure 4.12: MOS vs. number of calls

4.4.3. Results and analysis

Figure 4.11 shows the average E2E delay as a function of pause time for considered number of calls. The maximum speed of the nodes was set to 5 m/s and the pause time was varied as 0s, 10s, 15s, 20s and 25s. The E2E delay is decreasing for both protocols when there is little node mobility (i.e., at large pause time). In the presence of high mobility, link failures can happen very frequently which trigger new route discoveries and generate delay. Otherwise, E2E delay increases when the number of calls increases. However, our adaptive protocol permits to have less delay.

In Figure 4.12, we measure the average MOS versus the number of calls which is increased to see the impact on performance. We observe that both protocols give similar voice quality when the number of calls is fewer (less than 6 calls). This is expected as the network is not loaded. In this way, less bandwidth is wasted and less delay is introduced. From six calls a fair listening quality is shown for both protocols. However, for an acceptable voice quality delivered (MOS=3.6), adaptation accommodates more call capacity when compared to traditional, non-adaptive approaches. We can have call capacity until 10 calls with this score when applying the adaptive mechanism. This because OLSR-VA uses some of the non-loaded paths to deliver VoIP packets while OLSR uses some of the less stable ones and hence saturating them. In this case, our adaptation mechanism might increase calls capacity with trade-off between the achieved quality and the quantity of accepted calls.

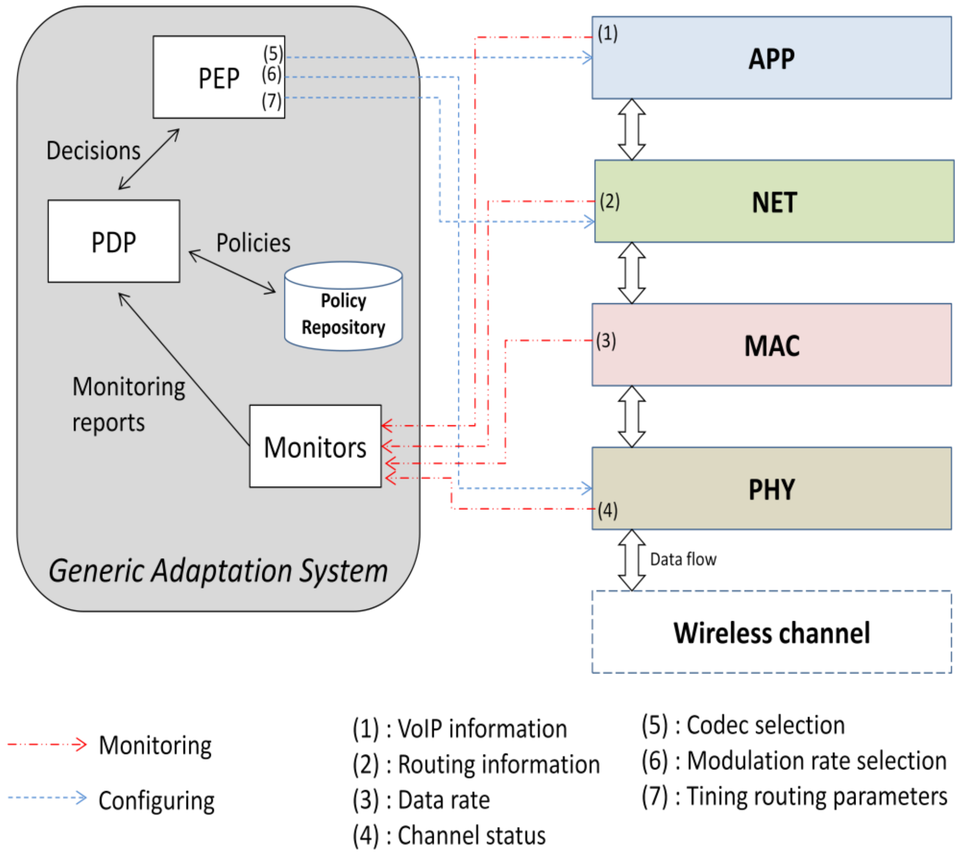


Figure 4.13: Generic adaptation system architecture

4.5. Toward a generic adaptation architecture for QoS management in VoMAN

In this section, we make the first step toward designing generic architecture that includes different QoS adaptation mechanisms (i.e. PHY/MAC, APP/MAC and NET/APP adaptations). We propose autonomous policy-based cross-layer architecture dedicated to QoS management in MANET carrying VoIP. Our objective is to introduce a degree of autonomy to the MANET, so as adapt the protocol's behaviour to network environment change.

Figure 4.9 shows the proposed generic adaptation system architecture. As is it shown, our solution is based on two main paradigms; the first is the cross-layering design technique and the other is the policy-based autonomic computing. The system function is expressed in terms of workflow (monitoring + configuration) or behavioural rules to be performed in response to certain situations. Next, we detail our architecture based on these paradigms.

4.5.1. Cross layer design

Several cross-layer architectures are designed to signalling or sharing information between different OSI layers. As presented in Subsection 2.4.2.3, our cross-layer design is inspired from the second category where a shared data-base provides of storage/retrieval information to all the layers (Recall Figure 2.14). In our system architecture the data-base is modelled as an autonomic system which collects the monitoring information from OSI layers, and configures parameters of the protocols implemented in different layers.

4.5.2. Policy-based autonomic computing

As reviewed in the Subsection 2.4.3.3 autonomic computing approach [56] combined with PBNM [52], provides the self-managing capabilities to network elements, allowing them to define what decisions should be taken in certain well-defined circumstances and possible environment changes. So, we adopt this paradigm in our solution by keeping PBNM terminology [52], but we attribute novel functionalities and roles to well-known entities. So, our basic components are:

- ***Monitor:*** is the component that gathers monitoring VoIP information and report them to VPDP. The collected information can be local, such as triggered events (i.e. local VoIP activity, routing performance) or external such as the signalling messages from neighbours (i.e neighbour VoIP activity). As shown in Figure 4.9, the monitoring can be made at deferent layers. For instance, monitor can collect information about data-rate, VoIP information, codec parameters, routing parameters and channel status.
- ***VPDP (VoIP Policy Decision Point):*** this entity makes decisions based on appropriate predefined policies and reported monitoring information collected by the monitor. Additionally, VPDP is considered as the final authority for the decisions that mast be enforced by VPEP. As example, if the monitor reports information about a changed data rate, the VPDP must find and process policies, then send decision to PEP in order to select the appropriate codec.
- ***VPEP (VoIP Policy Enforcement Point):*** this element enforces VPDP decisions by configuring deferent protocol parameters according to policies scope. As example, the VPDP can configure or reconfigure routing parameters, select a codecs and tune date rate at physical layer.

- ***Policy repository:*** this is a local database where policies are presented and stored in a structured way. In the next section, we describe briefly the policy representation.

Policies will be used to enable administrator high level goals to be automatically accomplished. So, they allow adapting system behaviours to network environment changes. They consist of a set of condition-action rules. However, policies specify what should happen in a certain situation and they do not precise how it should be done. For example, based on Policy Core Information Model (PCIMe) [120], we can define a policy as follow:

Policy ID:	Permit to identify the policy	e.g. P1
Policy Group:	Membership group	e.g. Group configuration
Policy Scope:	Represent it target	e.g. APP layer
Policy Event:	Event that trigger the policy execution	e.g. Changed VoIP codec
Policy Condition:	Boolean expression that specifies a well define situation	e.g. Sample size is high
Policy Action:	Specified what should be happen when condition are satisfied.	e.g. Tune Routing parameters

Finally, in this section we are just introduced an adaptive autonomous management system for VoMAN to allow mobile VoIP nodes to configure automatically their protocols parameters and to self-adapt to environment change. However, this architecture needs further investigation and implementation in order to test its performance. This should be a matter for an upcoming future work.

4.6. Conclusion

In this chapter, we have proposed a new adaptation mechanism integrated into OLSR routing protocol, which aims at improving VoIP application over MANET an acceptable quality. We have described the design of the various schemes involved. The solution was tested by deploying voice call services over high dynamic ad hoc networks (VANETs). The simulation was based on inter-vehicle voice streaming rely on multi-hop, and results are presented in terms of both network level (DVPQ) and user level (MOS) metrics. Based on investigating various simulation scenarios, results show that our solution (OLSR-VA) perform better than OLSR to deal with codecs change, and provide considerable improvement of VoIP quality.

Our realization efforts include a design of a generic QoS management architecture which incorporates adaptation mechanisms at different OSI layer. This architecture is characterized by the ability to self-adapt protocol parameters to network condition and changes.

Chapter 5

Conclusion & Future work

5.1. General conclusion

Mobile ad hoc network is an emerging field in networking area. Transmission of voice over such network makes it more applicable in real world. Moreover, Real-time voice communication over MANET is very much demanding and necessary, especially in emergency scenarios. However, VoIP requires real time access and some form of speed, more bandwidth and other resources to transfer it across the network. When taking into account MANETs characteristics and constraints, providing and managing QoS for VoMAN remains a challenge.

The last few years have witnessed small steps towards supporting voice and general multimedia over wireless and mobile ad hoc networks. This includes techniques which have been studied in the context of multimedia transmission over cellular networks, such as resource reservation, call admission control and link adaptation mechanism. Our work represents a first step toward improving aspects at the network layer making routing protocol aware of the voice travel and will improve the quality of the voice over MANETs.

This dissertation addresses VoMAN issues from the aspect of adaptation, claiming that effective adaptation of routing parameters can enhance VoIP quality. In the first step, we established a state of the art about the technical building blocks of the subject. We addressed three key elements surrounding VoIP technology, mobile ad hoc networks and quality of service management. In the second step, this work aimed at providing meaningful results to guide the design of an efficient solution. We have performed extensive simulation studies which have been highly beneficial to define parameters that need adaptation.

We studied the impact of tuning OLSR routing protocol parameters on VoMAN. This research has helped us understand the behaviour of VoIP codecs when varying OLSR parameter values. We showed that a quantitative relationship between VoIP codecs performance and factors like refresh intervals and willingness; hence, adaptation strategy and protocol is needed.

The most important contribution of the thesis is the adaptive OLSR-VA algorithm which provides an integrated environment where VoIP activity is constantly detected and the adaptation mechanism addresses it efficiently. The proposed routing adaptation algorithm is composed of two phases:

- *Monitoring* VoIP activity and codec change notifications using APP-layer for the local node or signalling messages for neighbour nodes.
- Adaptation, for choosing suitable routing configurations; this leads to bandwidth optimization by balancing VoIP load in the network and by adjusting signalling message overhead.

Motivated by the desire to minimize protocol complexity, we consider the proactive OLSR routing protocol (UM-OLSR ns-2 implementation) as a platform to develop the solution. We have extended the protocol by including adaptation functions and modifying heuristic routing. Furthermore, new HELLO message structure has been used for the dissemination of the VoIP activity information. This structure doesn't introduce any extra overhead as the unused parts of original HELLO messages are utilized.

To investigate the performance advantage achieved by such algorithm, we use concert simulation scenarios which provide a good cost effective environment. A number of realistic simulations under different conditions are performed (VANET and healthcare application scenarios). The most important observation is that performance is satisfactory, in terms of the perceived voice quality and call capacity. Results have shown ability of the solution for successfully achieve an acceptable voice quality even over long routes and under reasonably load conditions.

The proposed adaptation approach could be implemented complementarily with the other adaptation solutions. As presented in the last part of chapter 4, the proposed scheme can be integrated into a generic QoS adaptation management architecture. The architecture design is based on two paradigms; policy-based autonomic computing and cross layer design. This allows the system to be fully decentralized and the management process will be performed in an autonomous fashion.

5.2. Future directions

Providing high level QoS for multimedia service support over mobile ad hoc networks is an active research area. Our proposed solution seems highly beneficial for VoIP over MANETs. However, a number of avenues for further research remain open though. In the following we show how our work can be extended and expanded in various research directions.

- In the first we plan to continue working on the subject considering the impact of traffic heterogeneity, where voice, data and video are supported. In order to maintain good quality, we will elaborate the benefit of our solution on data applications which are delay-elastic, and intolerant to loss. This might be made by employing the measures of priority queuing and the percentage of data to voice traffic.
- The VoIP call capacity issue in MANET also deserves additional efforts. While typical scenarios in this thesis have been considered, other scenarios might be studied in order to show the trade-off between VoIP capacity and quality. Thus, we intend to exploit the cross layering to improve the performances of our algorithms. We can capture more information from different layers to optimize the communication protocols and VoIP activity scheduling and meet the application requirements.
- In Chapter 4, we have presented generic QoS management architecture design for VoMAN based on policies and cross layer architecture. We intend to implement the proposed architecture in ns-2 to further evaluate it performance and calculate the additional cost in terms of complexity, delay, computational power and storage incurred by the mechanism.
- We plan to further extend the proposed solution to full Wireless Sensor Networks implementation. However, another issue that is important here is energy consumption. The solution must be energy efficient that will extend network lifetime. For instance, the solution has to take into account energy as a metric and adapt it behaviour by selecting the most appropriate energy efficient strategies.
- As another matter of future works is the prototypical implementation of the whole mechanism which will help to validate the simulation results and calculate more precisely the additional cost. Additionally, we wish to test the deployment of our solution in larger scale environment.

In short, the work in this dissertation is merely the first steps in reaching the ultimate goal of implementing a management system with core QoS guarantees for VoIP applications over MANETs. The proposed solution represents a significant departure from the current research directions and warrants much future research.

Publications

Based on the results obtained from this thesis, the following publications were made:

- **In journals**

1. S. El Brak, M. Bouhorma, M. El Brak and A. Boudhir. "VoIP Application over MANET: Codec Performance Enhancement by Tuning Routing Protocol Parameters." Journal of Theoretical and Applied Information Technology (JATIT), Vol. 50, No. 1, April 2013. (SCOPUS indexed)
2. S. El Brak, M. Bouhorma, M. El Brak and A. Boudhir. "Speech Quality Evaluation Based Codec for VoIP over 802.11p." International Journal of Wireless & Mobile Networks (IJWMN), Vol. 5, No. 2, April 2013.
3. S. El Brak, M. Bouhorma, A. Boudhir. "VoIP over MANETS (VoMAN): QoS & Performance Analysis of Routing Protocols for Different Audio/Voice CODECS." International Journal of Computer Applications (IJCA), Vol. 36, N°12, pp. 22-26, November 2011.
4. S. El Brak, M. Bouhorma, A. Boudhir. "Network Management Architecture Approaches Designed for Mobile Ad hoc Networks", International Journal of Computer Applications (IJCA), Vol. 15, N° 6, pp. 14-18, February 2011.

- **In proceedings of International Conferences**

1. S. El Brak, M. Bouhorma, A.A. Boudhir, M. EI Brak, D. Benhaddou "Toward Making OLSR Aware of Voice Codec Configurations: Comprehensive Study" *in Proc. Of the 3rd International Conference on Information Systems and Technologies (ICIST'2013)*, Tangier, 2013.
2. S. El Brak, M. Bouhorma, A. Boudhir "VoIP over VANETs (VoVAN): A QoS Measurements Analysis of Inter-Vehicular Voice Communication in Urban Scenario", in *Proc*. Of the *IEEE International Conference on New Technologies, Mobility and Security (NTMS'12),* pp. 1 – 6, Istanbul, 2012. (published)
3. S. El Brak, M. Bouhorma, A. Boudhir, M. El brak, M. Essaaidi. "Voice over VANETs (VoVAN): QoS Performance analysis of different voice

CODECs in urban VANET scenario." In *Proc.* of the *IEEE International Conference on Multimedia Computing and Systems* (ICMCS'12), pp. 360 - 365, Tangier, 2012. (published)

4. S. El Brak, M. Bouhorma, A. Boudhir, "VoIP over MANETS (VoMAN): QoS Measurement Analysis for Different Audio/Voice CADECs", Apparue dans Proceeding de 3ème édition des Journées Doctorales en Technologies de l'Information et de la Communication (JDTIC), Tanger, 07 juillet 2011.
5. S. El Brak, M. Bouhorma, " Gestion et Monitorage des réseaux Ad hoc", Apparue dans Proceeding de Congrès Méditerranéen de Télécommunication (CMT), Casablanca, 18 Mars 2010.
6. S. El Brak, M. Bouhorma, "Gestion des Réseaux Mobile Ad hoc", Apparue dans Proceeding de Congrès International Scientifique de l'Ingénierie (CCII), Tétouan, 03 Mars 2010.

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Appendix

A. Discrete Event Simulator: ns-2

In our simulations, we use ns-2, network simulator 2 (Version 2.34) [99]. It is the most popular network simulator used by researchers. Ns-2 is an event-driven simulator originally derived from REAL network simulator in 1989 and has evolved over the past years. In 1995, its development was supported by DARPA through the Virtual InterNetwork Testbed (VINT) project. Currently the development is carried out by Information Sciences Institute (ISI) in California and is supported through DARPA and NSF.

Ns-2 is an open source discrete event simulator used by the research community for research in networking. This simulator provides support for simulation of TCP, routing, and multicast protocols over wired and wireless (local and satellite) networks. ns-2 implements the following features :

1. Router queue Management Techniques DropTail, RED, CBQ,
2. Multicasting
3. Simulation of wireless networks
    - Developed by Sun Microsystems + UC Berkeley (Daedalus Project)
    - Terrestrial (cellular, ad hoc, GPRS, WLAN, BLUETOOTH), satellite
    - IEEE 802.11 can be simulated, Mobile-IP, and ad hoc protocols
4. Traffic Source Behaviour- www, CBR, VBR
5. Transport Agents- UDP/TCP
6. Routing and ad hoc routing such as DSR, TORA, DSDV and AODV.
7. Packet flow
8. Network Topology
9. Applications- Telnet, FTP, Ping
10. Tracing Packets on all links/specific links

The core ns-2 is written in C++ which is used to extend the simulator (i.e. to define protocol behaviours). OTcl (object-oriented Tcl) which is used for writing simulation scripts and manipulating of existing C++ objects, periodic or triggered actions, etc. As shown in Figure 1, While composing a simulation scenario, one needs to construct a binding between OTcl and the actual C++ classes by using a special bind procedure. Any modification to the objects initiated from one language is visible in the other. This allows access to the objects from either language and makes it easy to move functionality between these two programming realms.

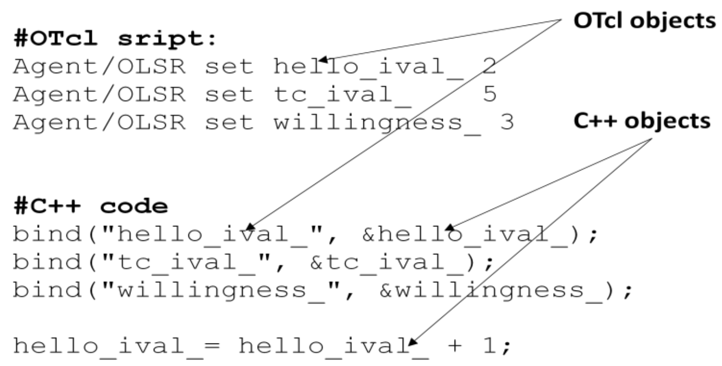


Figure 1. Object Binding between OTcl and C++ in ns-2

A simplified user's view of ns-2 is shown in Figure 2. The OTcl script is used to initiate the event scheduler, set up the network topology, and tell traffic source when to start and stop sending packets through event scheduler. The scenes can be changed easily by programming in the OTcl script. When a user wants to make a new network object, he can either write the new object or assemble a compound object from the existing object library, and plumb the data path through the object. This plumbing makes ns-2 very powerful.

The choice of languages involves a trade-off between performance and ease of use. OTcl makes it easier to rapidly prototype new simulation script, while C++ is more suitable for large simulations that require high performance. Therefore, functionality that requires per-packet processing in ns-2 should be implemented in C++, while the experimental code fragments that are not frequently used can be implemented in OTcl. Since this study is on routing protocols, we use C++ in implementation.

In our simulations we use ad-hoc networking extensions provided by the Rice (CMU Monarch) Monarch project [106]. In these extensions, each mobile node is an independent entity that is responsible for computing its own position and velocity as a function of time. Each mobile node can have one or more network interfaces, each of which is attached to a channel. Figure x shows the basic schematic layout of a typical mobile node.

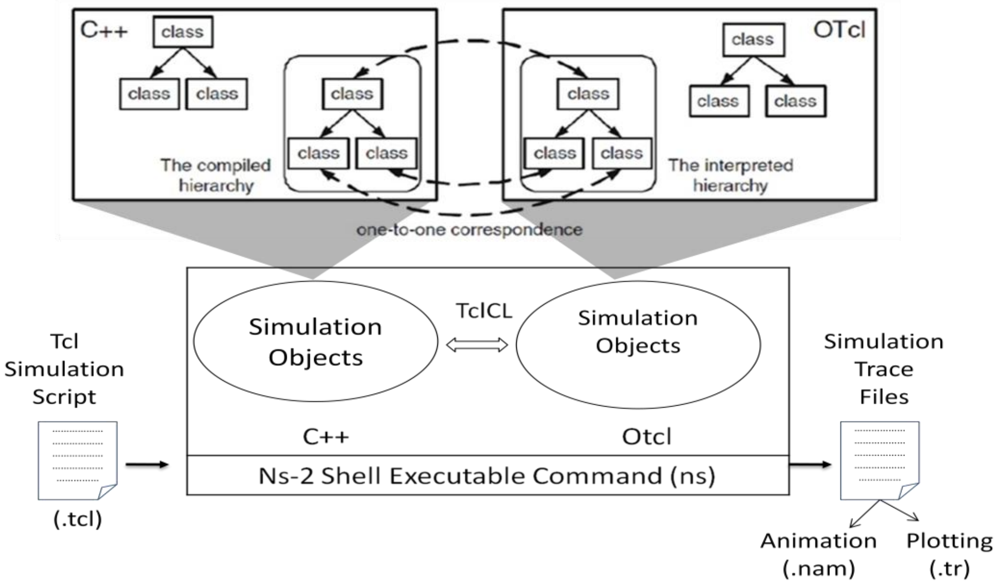


Figure 2. Architecture and user view of ns-2

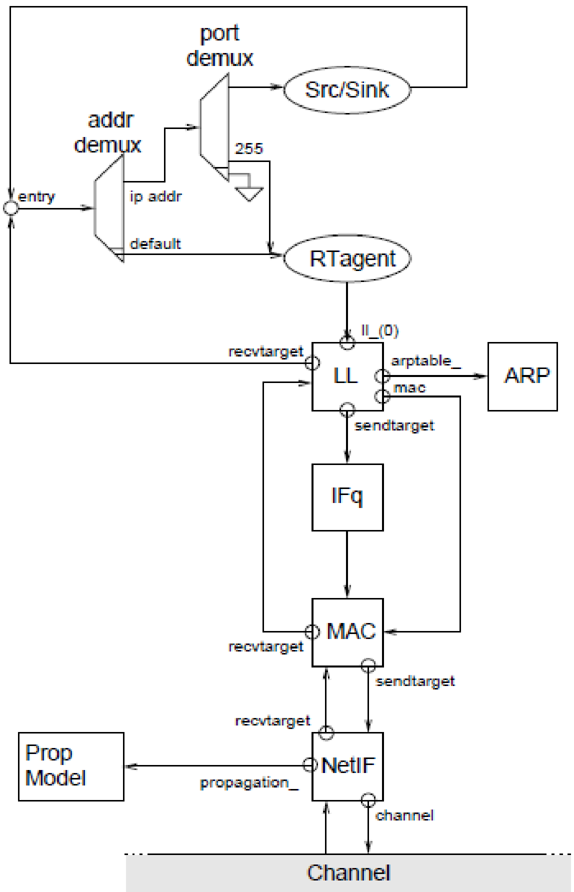
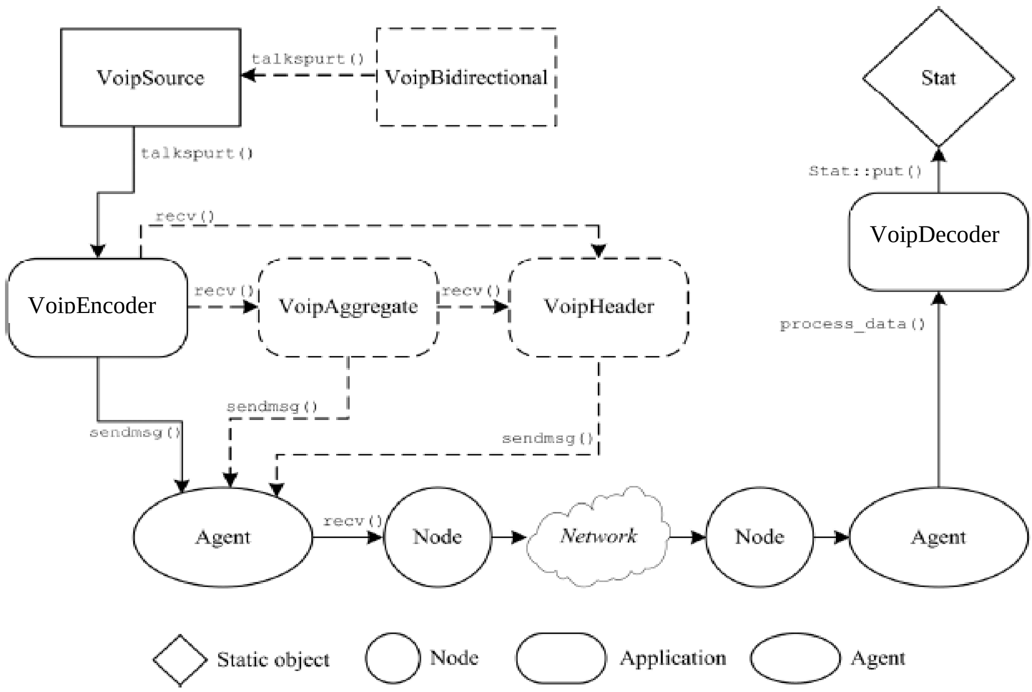


Figure 3. Schematic of a mobile node [106]

B. VoIP simulation in ns-2

- **ns2voip++ framework: Patch**

ns2voip++ [104] is a ns-2 extension allows to simulate voice traffic and evaluate VoIP application performance. This framework has the following modular architecture:



- ➔ Assume that you have downloaded and unpacked the ns-2 version 2.34 (let us assume the location is /usr/local).
- ➔ Copy the patch "ns2voip++.patch" in the directory: /usr/local/ns-allinone-2.34/ns-2.34/tmp
- ➔ Apply the patch, by executing the following command:

```
cd /usr/local/ns-allinone-2.34/ns-2.34
cat /tmp/ns2voip++.patch | patch -Np1
```

- ➔ If you have not yet installed ns-2, follow these steps:

```
cd ..
./install
```

➔ Else, if you install *ns2voip++* on an older ns-2 installation, recompile ns-2.

```
./configure
make distclean
./configure
make
```

- **ns2voip++ framework: interface**

Know, we will configure *ns2voip++* settings at transmitter and receiver. This requires:

- **Transmitter side :**

➔ Create *VoipSource* and *VoipEncoder* objects, and then attaches *VoipEncoder* object to a UDP agent.

```
set voip_source [new VoipSource]
set voip_encoder [new Application/VoipEncoder]
set agt_snd [new Agent/UDP]
$voip_source encoder $voip_encoder
$voip_encoder attach-agent $agt_snd
```

➔ Run / stop talkspurts generation

```
$voip_source start|stop
```

➔ Specify the speech model

```
$voip_source model exponential $on $off |one-to-one |one-to-many
|many-to-one |many-to-many
```

➔ Select the codec to use for encode the VoIP frames. Four popular codecs are implemented. In addition you can add other (modification of C ++ code is needed).

```
$voip_encoder  codec G.711 | G.729.A |GSM.AMR | G.723.1 | GSM.EFR
```

➔ Configure aggregation object so that all packets ns-2 contains $n VoIP frames.

```
$voip_aggr nframes $n
```

➔ Set the size of the IP / UDP / RTP header.

```
$voip_compression $size | nocompression
```

- **Receiver side:**

➔ Create decoder application (*VoipDecoder*) and attach it to a UDP agent

```
set voip_decoder [new Application/VoipDecoder]
set agt_dst [new Agent/UDP]
$voip_decoder  attach-agent $agt_dst
```

➔ Associate the numeric identifier $ID with the decoder (*VoipDecoder*)

```
$voip_decoder id $ID
$voip_decoder cell-id $cell
```

➔ Configure E-model parameters (`$ie $bpl $a $ro $th`)

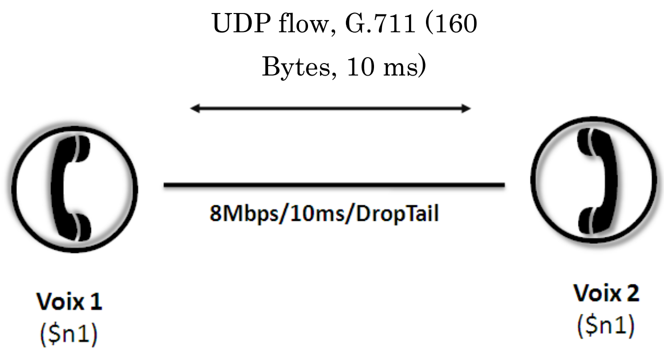
```
$voip_decoder emodel $ie $bpl $a $ro $th | G.711 | G.729 | GSM.AMR
```

➔ Specify the playout buffer parameters.

```
$voip_decoder buffer-size $n
$voip_decoder initial-delay $delay
$voip_decoder playout-rate $time
```

- **ns2voip++ framework: Simulation example**

We will simulate a single VoIP network installation, corresponding to the following diagram:



We will create two nodes ns-2 (n), connected by a full duplex link. First, create a TCL file to simulate VoIP using ns2voip Framework. Given below the description of the parts of source code:

➔ Configuring global parameters (simulation time, the type of codec, and speech model)

```
set opt(duration)      100.0          ;# Simulation time
set opt(voip-model)    one-to-one     ;# VoIP VAD model
set opt(voip-codec)    G.711          ;# VoIP codec
```

➔ Creating a procedure to create bi-directional VoIP source

```
proc create_voip { fid start stop } {
   global ns voip
   set app [new VoipSource]
   $app model $opt(voip-model)
   set bidirectional [new VoipBidirectionalModifiedBrady]
   $ns at $start "$bidirectional start"
   $ns at $stop "$bidirectional stop"
   $bidirectional source $app
   $app bidirectional $bidirectional
   set encoder [new Application/VoipEncoder]
   $encoder id $fid
   $encoder codec $opt(voip-codec)
```

```
   $app encoder $encoder
   set decoder [new Application/VoipDecoderOptimal]
   $decoder emodel $opt(voip-codec)
   $decoder id $fid
   $decoder cell-id 0
   set voip(encoder) $encoder
   set voip(decoder) $decoder
}
```

➔ Creating a procedure to create two UDP agents and attach them to: encoder and decoder.

```
proc create_udp { n0 n1 fid } {
   global ns voip
   set agtsrc [new Agent/UDP]
   set agtdst [new Agent/UDP]
   $agtsrc set fid_    $fid
   $ns attach-agent $n0 $agtsrc
   $ns attach-agent $n1 $agtdst
   $ns connect $agtsrc $agtdst
   $voip(encoder) attach-agent $agtsrc
   $voip(decoder) attach-agent $agtdst
    }
```

➔ Simulator instance

```
set ns [new Simulator]
set tr [open voip_trace.tr w]
$ns trace-all $tr
set namtrace [open voip_nam.nam w]
$ns namtrace-all $namtrace
```

➔ "finish" procedure

```
proc finish {} {
       global ns namtrace tr
       $ns flush-trace
       close $tr
       close $namtrace
       exec nam voip_nam.nam &
```

```
        exit 0
        }
```

➔ Nodes creation

```
set n0 [$ns node]
set n1 [$ns node]
set fid 1
set start 0
set stop 95
$n0 color red
$n1 color blue
$n0 label "Voix 1"
$n1 label "Voix 2"
$ns color 1 red
$ns color 2 blue
```

➔ Create a duplex link between nodes.

```
$ns duplex-link $n0 $n1 8Mb 50ms DropTail
$ns duplex-link-op $n0 $n1 orient right
```

➔ Creation of two VoIP streams (stream 1 between n0 and n1, and stream 2 between n1 and n0)

```
for { set i 0 } { $i < 1 } { incr i } {
      create_voip $fid $start $stop
      create_udp $n0 $n1 $fid

#créer le flux corrélée de sens opposé
      create_voip [expr $fid+1] $start $stop
      create_udp $n1 $n0 [expr $fid+1]
```

➔ `"finish"` procedure call

```
$ns at $opt(duration) "finish"
```

➔ Simulation run

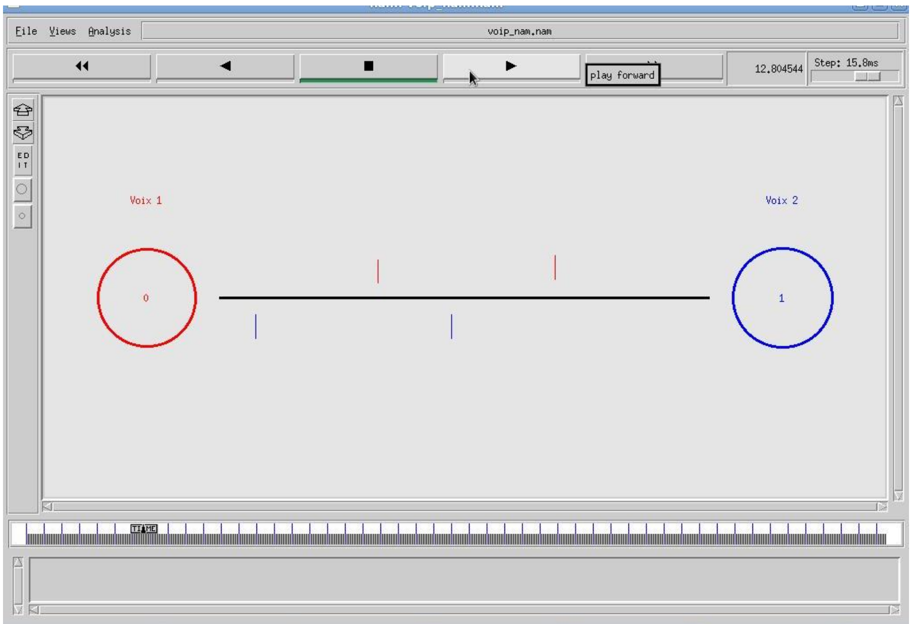
```
$ns run
```

➔ Simulation execution

```
ns simulation_voip.tcl".
```

➔ The file ".nam" produces motion graphics for the simulation. The result of the execution looks like this:

NB. You must have VoIP flows in both directions.



➔ The trace file ".tr" is used to analyze the results of our simulation.

C. QoS metrics measure: Awk scripts

- AWK script to measure average end to end delay

```
BEGIN {
       highest_packet_id =0;
       sum=0;
       recvnum=0;}
 {
  time = $3;
  packet_id = $41;
# CALCULATE DELAY
 if ( start_time[packet_id] == 0 )  start_time[packet_id] = time;
 if (( $1 == "r") &&  ( $35 == "udp" ) && ( $19=="AGT" )) {
end_time[packet_id] = time;  }
       else {  end_time[packet_id] = -1;  }
#find the number of packets in the simulation
 if (packet_id > highest_packet_id)highest_packet_id = packet_id;
  }
   END {
  for ( i in end_time ) {
            start = start_time[i];
            end = end_time[i];
            packet_duration = end - start;
  if ( packet_duration > 0 )
  {    sum += packet_duration;
       recvnum++;
  } }
     delay=sum/recvnum;
     printf delay*1000;
printf ("\n"); }
```

- AWK script for measure the throughput

```
BEGIN {
       recvdSize = 0
       startTime = 1e6
       stopTime = 0
     swh=0}
  {
```

```
# Trace line format: new
       if ($2 == "-t") {
             event = $1
             time = $3
             node_id = $5
             flow_id = $39
             pkt_id = $41
             pkt_size = $37
             flow_t = $45
             level = $19
       }

  # Store start time
  if ((level == "AGT") && (event == "s") && (pkt_size >=
pkt_size)) {if (time < startTime) {
             startTime = time
            swh=pkt_size}
       }
# Update total received packets' size and store packets arrival
time
  if (level == "AGT" && event == "r" && pkt_size >= swh) {
       if (time > stopTime) {
             stopTime = time }
 # Rip off the header
       hdr_size = pkt_size % swh
       pkt_size -= hdr_size
       # Store received packet's size
       recvdSize += pkt_size}
  }
  END {
#printf("Average Throughput[kbps] = %.2f\t\t
StartTime=%.2f\tStopTime=%.2f\n",(recvdSize/(stopTime-
startTime))*(8/1000),startTime,stopTime)
printf((recvdSize/(stopTime-startTime))*(8/1000)"\n")   }
```

- AWK script to measure packets loss.

```
  BEGIN {
       sends=0;
       recvs=0;
```

```
       droppedPackets=0;
     }
     {
  time = $3;
  # SEND  PACKETS
  if (( $1 == "s") &&  ( $35 == "udp" ) && ( $19=="AGT" )) {
sends++; }
   # DROPPED  PACKETS
  if (( $1 == "d" ) && ( $35 == "udp" )  && ( $3 > 0 ))
{droppedPackets++;}
     }
 END {
          printf (droppedPackets / sends)*100;printf ("\n");
  }
```

- AWK script for compute delay jitter

```
  BEGIN {
         num_recv=0 }
     {
   # Trace line format: new
       if ($2 == "-t") {
             event = $1
             time = $3
             node_id = $5
             flow_id = $39
             pkt_id = $41
             pkt_size = $37
             flow_t = $45
             level = $19
       }
     # Store packets send time
 if (level == "AGT" && sendTime[pkt_id] == 0 && (event == "s") &&
pkt_size >= 48) {
   sendTime[pkt_id] = time  }
# Store packets arrival time
  if (level == "AGT" && event == "r" && pkt_size >= 48) {
             recvTime[pkt_id] = time
             num_recv++  }
  }
```

```
END {
     # Compute average jitter
     jitter1 = jitter2 = tmp_recv = 0
     prev_time = delay = prev_delay = processed = 0
     prev_delay = -1
     for (i=0; processed<num_recv; i++) {
           if(recvTime[i] != 0) {
                   tmp_recv++
                if(prev_time != 0) {
                     delay = recvTime[i] - prev_time
                     e2eDelay = recvTime[i] - sendTime[i]
                     if(delay < 0) delay = 0
                     if(prev_delay != -1) {
                     jitter1 += abs(e2eDelay - prev_e2eDelay)
                     jitter2 += abs(delay-prev_delay)
                     }
                     prev_delay = delay
                     prev_e2eDelay = e2eDelay
                }
                prev_time = recvTime[i]
           }
           processed++
     }
}

END {
       printf("Jitter1 = %.2f\n",jitter1*1000/tmp_recv);
       printf("Jitter2 = %.2f\n",jitter2*1000/tmp_recv);
}
   function abs(value) {
     if (value < 0) value = 0-value
     return value
}
```

- AWK script to measure normalize routing load,

```
BEGIN {
     recvs=0;
     routing_packets=0.0;
   }
```

```
     {
# CALCULATE PACKET DELIVERY FRACTION
if (( $1 == "r") &&  ( $35 == "udp" ) && ( $19=="AGT" ))    {
recvs++; }
 # CALCULATE TOTAL DSR OVERHEAD
  if (($1 == "s" || $1 == "f") && ($19 == "RTR") && ($35
=="OLSR") ){routing_packets++;}
     }
  END {
     NRL = routing_packets/recvs;  #normalized routing load
     printf NRL; printf ("\n");
  }
```

- AWK script to measure: send, receive, routing, drop packets, Packet delivery ratio, routing overhead, normalize routing load

```
 BEGIN {
       sends=0;
       recvs=0;
       routing_packets=0.0;
       droppedBytes=0;
       droppedPackets=0;
       highest_packet_id =0;
       sum=0;
       recvnum=0;
     }
  {
  time = $3;
  packet_id = $41;
  # CALCULATE PACKET DELIVERY FRACTION
  if (( $1 == "s") &&  ( $35 == "udp" ) && ( $19=="AGT" )) {
sends++; }
  if (( $1 == "r") &&  ( $35 == "udp" ) && ( $19=="AGT" ))    {
recvs++; }
   # CALCULATE TOTAL ROUTING OVERHEAD
  if (($1 == "s" || $1 == "f") && $19 == "RTR" && $35 =="OLSR")
routing_packets++;
   # DROPPED ROUTING PACKETS
  if (( $1 == "d" ) && ( $35 == "udp" )  && ( $3 > 0 ))
       {
```

```
              droppedBytes=droppedBytes+$37;
              droppedPackets=droppedPackets+1;
        }
 #find the number of packets in the simulation
           if (packet_id > highest_packet_id)
              highest_packet_id = packet_id;
  }
END {
        NRL = routing_packets/recvs;  #normalized routing load
     PDF = (recvs/sends)*100;  #packet delivery ratio[fraction]
     printf("----------\n");
     printf("send = %.2f\n",sends);
     printf("recv = %.2f\n",recvs);
     printf("drop = %d\n",droppedPackets);
     printf("routingpkts = %.2f\n",routing_packets++);
     printf("dropped data (bytes)   = %d\n",droppedBytes);
     printf("----------\n");
     printf("PDF(%) = %.2f\n",PDF);
     printf("----------\n");
     printf("Packet loss(%)=%.2f\n",(droppedPackets/sends)*100);
     printf("----------\n");
     printf("NRL = %.2f\n",NRL);
     printf("----------\n"); }
```

Comments

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