VoIP over MANET (VoMAN): QoS & Performance Analysis of Routing Protocols for Different Audio Codecs

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ABSTRACT
Voice over IP (VoIP) has become a popular Internet application. Mobile Ad hoc Networks (MANETs) may provide a good platform for the fast deployment of VoIP service in many application scenarios. The luck of infrastructure, flexibility and low cost are the main characteristics of MANETs. Otherwise, its present a considerable complexity that makes the transmission of real-time applications like VoIP a great challenge due to Quality of Service (QoS) requirements. This paper investigates the performances of routing protocols (AODV, OLSR) in MANETs carrying VoIP traffic. Via a simulation study we analyze and evaluate some QoS indicators like bandwidth, end-to-end delay and packet loss. Using Network Simulator (ns2), several voice codecs are studied to determine their effect on metrics QoS. We show how these codecs affect the performance of the routing protocols when varying hops number.

General Terms
Real traffic, Voice over IP, Ad hoc Networks.

Keywords
MANETs, VoIP, QoS, CODECs, NS2

1. INTRODUCTION
MANETs are autonomous networks consisting of two or more mobile nodes equipped with wireless communication and networking capabilities, but they don't have any network centralized infrastructure [11]. Routing protocols defines how packets will be delivered. In this work we use two IETF standard routing protocols, reactive AODV [5] and proactive OLSR [4].

Voice over Internet Protocol (VoIP) is a technology that allows you to make voice calls using an Internet connection instead of a regular (or analog) phone line. The QoS on VoIP network partly depends on the types of voice codec used [2]. The primary functions of a voice codec are to perform analog/digital voice signal conversion and digital compression. These codecs differ in their coding rate (bps), frame rate (frames/s), algorithmic latency that will influence the speech quality in a VoIP network.

In this paper, we will focus on investigating MANETs performance in a VoIP Context. Network Simulator 2 (ns2) is used to run several simulations, we use ns2voip++ [6] module to generate voice traffic. We make a measurement on VoIP channel characteristic such as delay, bandwidth, packet loss contributing to QoS for varying hops number and routing protocols with four different codecs which are G.711, G.723.1, G.729 and GSM.AMR.

This paper is organized as follows: Section 2 is the introduction to the VoIP which provides information of the technical aspect of VoIP such as protocol stack, coding and traffic. Section 3 presents QoS parameters in Viop Service. Section 4 shows the methodology for the simulation which uses the ns2 as simulation tools. Results and analysis are presented in section 5. Then we conclude this paper.

2. VOICE OVER INTERNET PROTOCOL
Figure 1 describes a VoIP system [12]. Normally, the speech source alternates between talking and silence period, which is typically considered to be exponentially distributed. Before transmitted over packet switched networks, the speech signal has to be digitized at the sender; the reverse process is performed at the receiver.

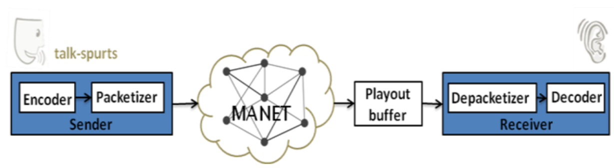

Fig 1: VoIP system

The digitalization process is composed of sampling, quantization and encoding. There are many encoding techniques that have been developed and standardized by the ITU (International Telecommunications Union) [2] such as G.711, G.729 and G.723.1. The objective of a codec is to obtain the lowest rate bit stream possible after conversion without degrading the quality of the signal. Table 1 describes various codecs [1] used on packet networks. The bit rate depends on the codec used and is the number of bits per second required to deliver a voice call.

The sample size is the number of bytes captured from the analog signal during the sample interval. Packets Per Second (PPS) represents the number of packets that must be transmitted in order to maintain the codec bit rate. The payload size is the bytes that fill the packet.

Table 1. Voice codecs used in packets networks.

Codec	Bit rate (kbps)	Sample size (bytes)	Packets per second	Payload size (bytes)
G.711	64	80	50	160
G.723.1	6.3	24	34	20
G.726.A	32	20	34	80
GSM	13.2	20	50	33

The encoded speech is then packetized into packets of equal size. Each such packet includes the headers at the various protocol layers such Real-time Transport Protocol (RTP) 12 bytes, User Datagram Protocol (UDP) 8 bytes, Internet Protocol version 4 (IPv4) 20 bytes and the payload comprising the encoded speech for a certain duration depends on the codec deployed (Figure 2).

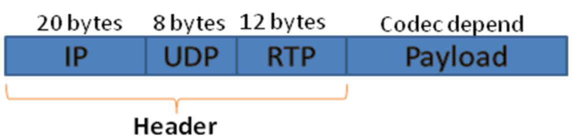

Fig 2: VoIP Packet

As the voice packets are sent over IP networks and wireless channel, they incur variable delay and possibly loss. In order to provide a smooth playout delay, at the receiver, a playout buffer is used to compensate the delay variations. Packets are held for a later playout time in order to ensure that there are enough packets buffered to be played out continuously [10].

3. ROUTING PROTOCOLS

Routing protocols defines how packets will be delivered. In this section we present two IETF standard routing protocols, reactive AODV and OLSR.

3.1 AODV

AODV (Ad Hoc On-Demand Distance Vector – IETF RFC 3561) [14] is a reactive routing protocol, so routes are created only when they are needed. Each host maintains a routing table which stores the next hop information to the destination and a destination sequence number which indicates the last known route to the wanted host. Route discovery is done by broadcasting a request message RREQ to the neighbors with the destination sequence number in order to prevent old information to be replied and to prevent loops.

Every host that receives RREQ and it is not the destination or does not know any route to it, increments its hop metric and updates its routing table. This procedure helps the route reply message RREP to be routed back to the requesting host. Any host, and not only the destination itself, can respond to a route request if it has an active route to that host unless RREQ bit D (Destination only) is set.

When a link fails, e.g. by nodes motion, an error message RERR is sent to all affected nodes. The affected nodes are known due to the precursor list which each node maintains and contains a list of nodes which use this node as route to any other. Failure detection can be done in two ways: by the absent of correct hello messages from neighbors or by means of information fromMAC layer. Hello messages are exchanged in order to nodes know their neighbors, when a hello is sent and it is not received correctly, nodes assume that there is a link failure.

3.2 OLSR

OLSR (Optimize Link State Routing - IETF RFC 3626) [15] is a proactive link state routing protocol. As a proactive protocol, OLSR constructs and constantly maintains information about network topology by means of exchange link state information. Each OLSR node sends HELLO messages in predefined time intervals for constructing its 1-hop and 2-hop neighbor sets and a TC (topology control) message for completing link state information, so routing table can be calculated. Link failures in OLSR are detected this way.

OLSR introduces multipoint relays (MPRs) in order to reduce message overhead in network. The MPR set of a given OLSR node is a subset of its neighbors which can forward its control messages. The neighbors which a given node A selects as MPR are called MPR nodes of A. When all neighbors are MPR nodes of a given router, OLSR diffuses control messages similarly to classical flooding mechanism. On the other hand, MPR mechanism described in [15] can decrease network performance due overhead introduced for constructing and repairing MPR set. If one or more MPR nodes fail, link state information can not be completely diffused, thus some routers can forward user data by invalid paths on network.

4. QOS PARAMETERS IN VOIP SERVICES

Performance metrics indicators for QoS are used to establish the performance of systems. The performance metrics are delay, packet loss and bandwidth.

Bandwidth

Throughput is the total number of bits that are sent through the channel per second. The channel is the ad hoc network, thus, bandwidth is the maximum number of bits that can be sent per second through the ad hoc network.

End-to-End Delay

Delay is measured from the instant a packet leaves the sender's Network Interface Card (NIC) to the instant it is received at the destination's NIC. According to ITU Recommendation G.114, delay in VoIP applications should never exceed 400 ms otherwise the quality of the VoIP stream is significantly degraded. However, the average delay for a VoIP stream should be less than 150 ms for acceptable perceived quality [7]. This end-to-end delay includes any time needed to calculate a new route and other routing delays such as router (i.e., another ad hoc node) processing and queuing delays.

Packet Loss

VoIP applications are sensitive to packet loss. Even though VoIP applications tolerate packet loss up to 10%, a packet loss of 1% still affects the quality of the VoIP stream [9,8]. Packet loss is measured as the percent of packets dropped at the receiver prior to data stream playback.

5. SIMULATION

Measurement of an actual MANET is expensive and infeasible. Therefore, the evaluation technique is simulation; we have used the discrete event network simulator ns2.34 [13]. The system under test is the VoIP MANET (VoMAN) System. Table 2 shows the simulation parameters that affect the performance of the VoMAN system. The simulation scenario consists of 50 nodes in an area of 1000x1000m2 created with random placement (Figure 3). VoIP traffic is introduced into the network with ns2voip++[1] module that has been integrated to ns2. To simulate and analyze performance of OLSR protocol, UM-OLSR[2] implementation is installed too.

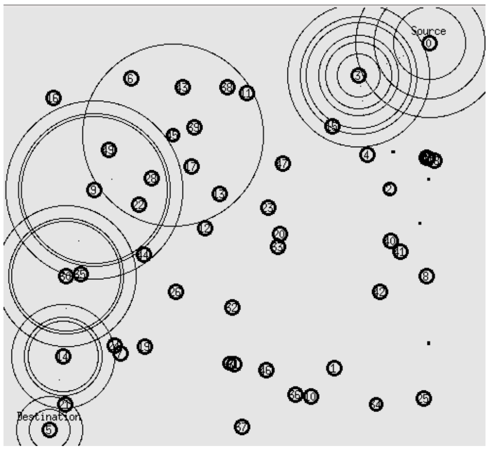

Fig 3: Network topology

Using IEEE 802.11 for the MAC layer, one VoIP streams is sent across the network from node source to node destination. For every routing protocol, four codecs have been simulated with duration of 100 seconds. We examine whether hops affect VoIP performance over MANETs.

Tab. 2: Simulation parameters

Parametres	Value
Routing protocol	OLSR/AODV
Mac/Phy	802.11
Area	1000m2
Traffic type	VoIP
Codecs	G.711/G.729/G.723.1/GSM
Number of nodes	50
Nodes position	Random
Queue type	/DropTail/PriQueue
Simulation time	100s

6. RESULTS & ANALYSIS

In the following, we show and discuss our simulation results investigating the impacts of the introduced QoS mechanisms. QoS will be measured in terms of bandwidth, packet loss and delay. Here we are trying to transmit voice data over wireless multi-hop network, and testing how hops affect the quality of the voice.

6.1 Bandwidth

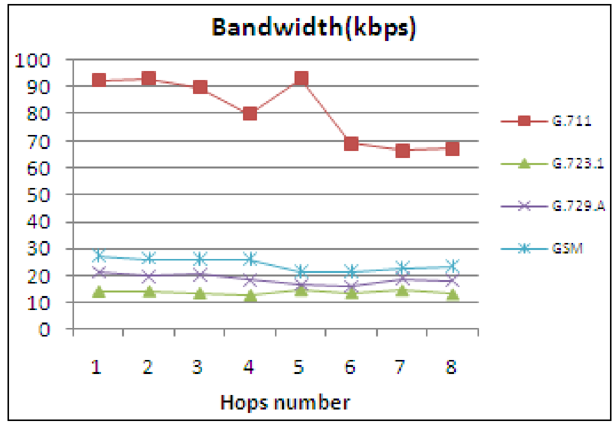

Fig 4: AODV bandwidth

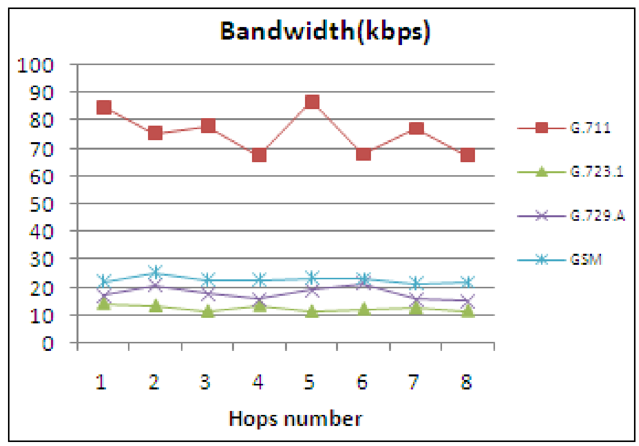

Fig 5: OLSR bandwidth

Figures 4, 5 illustrate the bandwidth measured versus hops number for different codecs. As can be observed G.711 gives the highest bandwidth versus other codecs, the same behavior observed for reactive (AODV) and proactive (OLSR) protocols. The low bandwidth of other codecs behind G.711 codec can be explained by small voice packets and sent very frequently.

6.2 Delay

Figures 6 and 7 show the end-to-end delay, delay is the lowest when hops are less than 4 for AODV protocol and less than 3 for OLSR protocol, as the hops number increase delay increase. But reactive protocol presents more delay than the proactive protocol. This is due to the reactive AODV route maintenance generating an increase in transmission queues when the topology changes.

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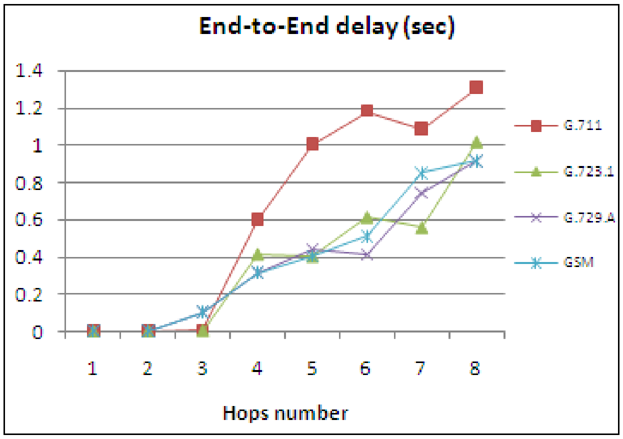

Fig 6: AODV Delay

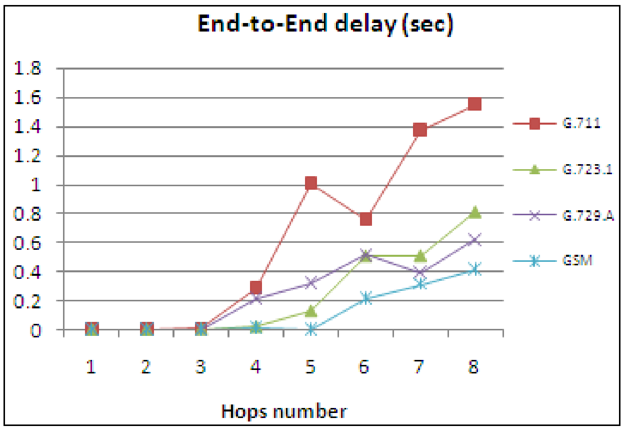

Fig 7: OLSR Delay

6.3 Packet loss

Packet loss is expressed as a ratio of the number of packets lost to the total number of packets transmitted. Packet losses results when packets sent are not received at the final destination. The percentage of packet loss increase for different coding technique at certain hops number (4 for OLSR and 3 for AODV).

For the simulation analysis, G.711 suffers dramatically from the packet loss compare with others codecs. 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.

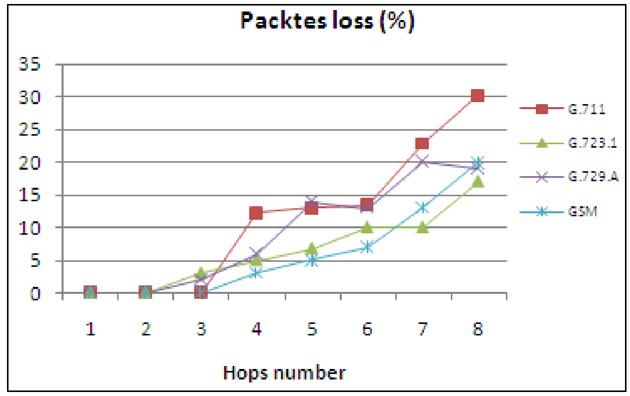

Fig 9: AODV Packet loss

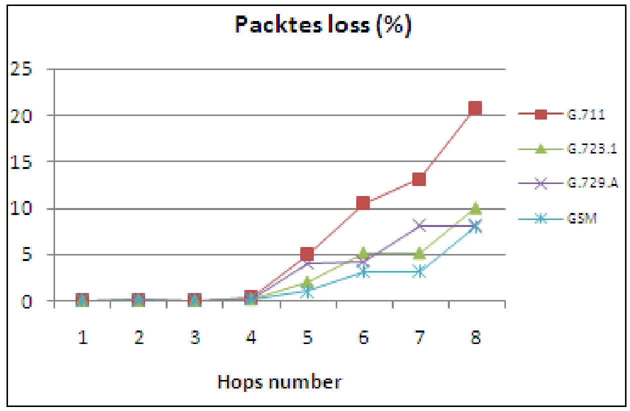

Fig 10: OLSR Packet loss

7. CONCLUSION & FUTURE WORKS

Ad-Hoc network is an emerging field in networking area. Transmission of voice over such network makes it more applicable in real world. In this paper we investigate how voice over IP (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 behavior in end-to end delay especially with GSM codec. 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.

The simulation studies proved that G.711 codec has more in and out traffic as its bandwidth consumption is highest among all the codecs. The delay performance of GSM is good comparatively to other codecs. GSM has the ability to provide the highest capacity for VoIP and has some features to deal with packet-loss.

In future work, first we intend to evaluate other routing protocols in a fort mobility environment. Second, we plan to design QoS management policies in order to solve the QoS problem when voice coexists with other traffic.

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