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VOIP APPLICATIONS OVER MANET: CODEC PERFORMANCE ENHANCEMENT BY TUNING ROUTING PROTOCOL PARAMETERS

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ABSTRACT

Abstract—Voice over Internet Protocol (VoIP) may provide good services through Mobile Ad hoc Networks (MANets) platform. It may cover many application scenarios range from safety to comfort related services. However, these networks introduce many challenges for supporting voice with QoS requirements. In this study, we take the first step in addressing the practical issues related to deploying voice call services over MANet. The approach is based on voice streaming between nodes rely on multi-hop by means of simulation. For this task, a performance evaluation of various voice codecs and its impact on quality of service metrics will be analyzed. Moreover, we study the impact of routing protocol parameter settings on codecs performance. The quality of the voice transmission, using different codecs, is measured and presented in terms of both network and user level metrics. This performance evaluation showed that selecting codec combined with tuning routing parameters, lead to optimal call quality.

Keywords: *MANet, VoIP, QoS, Codecs, Tuning OLSR, ns-2.*

1. INTRODUCTION

Voice over Internet Protocol (VoIP) is one of the most important technologies that allow making voice calls through Internet connection. As it is well known, the quality of service is important for VoIP applications. They especially require limited end-to-end delay and a low packet loss rate. To ensure quality voice communication, a suitable voice coding techniques (codec (i.e. coder/decoder)) are needed, which the primary function is to perform analog/digital voice signal conversion and digital compression [1].

In the last few years, mobile ad hoc networks (MANETs) have gained popularity as a method opens the way for a large range of applications like conferencing, emergency services, wireless sensor networks and vehicular ad hoc networks. However, such networks introduce several constraints like the high mobility of the nodes, frequently changing topology, hard delay, etc [2]. These make the transmission of multimedia traffic over such networks a challenging task. Routing protocols are important issue in mobile ad hoc networks, since there is no central manager entity in charge of finding the routing path among the nodes. Thus a great deal of effort is dedicated to design efficient routing protocols.

In this study, we focus on VoIP transmission over MANet by testing the quality of the voice. More specifically, the target of our paper is to study the performance of VoMAN (Voice over MANet) transmission when using different codecs and tuning routing parameters. For this task, we perform number of realistic MANet simulations under different conditions.

The main contributions of this work are:

- Generating several MANet instances following real data to achieve accurate simulations, Studying different QoS metrics to evaluate codecs performances,
- Testing how routing parameters can influence the human perceived quality of VoIP call, and deduct the tradeoff between tuning routing parameters and codec bite-rate.

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The rest of the paper is organized as follows: Section 2 overviews voice transmission over IP and MANets. Section 3 describes in detail the basic features of simulation methodology. Results and analysis are presented in section 4. In section 5 we overview related work. Finally, conclusions and future work are drawn in Section 6.

2. VOICE OVER IP OVER MANET

Figure 2 reports the different block diagram involved in a communication in a VoIP system [1]. Normally, the speech source alternates between talking and silence period, which is typically considered to be exponentially distributed. The speech will enter to the digitalization process that is composed of sampling, quantization and encoding. The encoded speech is then packetized into packets of equal size preparing them for transmission over IP network requiring appropriate signaling protocols such as H.323 [3] or SIP [4]. In the receiver side, encoded speech will be comprised by the playout buffer for certain duration depending on the codec deployed, than reverse process is performed (depacketized and decoded).

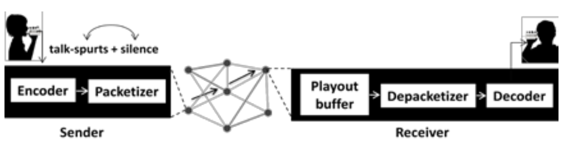

Figure 1. Voip System Block Diagram

The first step for voice communication is the application of a voice codec process, whose primary functions are to perform voice digitalization (analog/digital signal conversion) and compression (provides the lowest bit-rate streaming possible without degrading the quality of the signal). Various encoding techniques have been developed and standardized by the International Telecommunications Union (ITU-T). Table 1 shows some of the commonly used ITU-T standard codecs, and lists their attributes. Generally, the mean difference between them lies in the type of compression used. Encoders generate constant bit-rate audio frames consisting of 40 bytes IP/UDP/RTP headers followed by a relatively voice payload size.

TABLE I. ITU-T VOICE CODECS DESCRIPTION

Codec	*Sample size (bytes)*	*Sample interval (ms)*	*Bit-rate (kbps)*	*Payload size (bytes)*
G.711	80	10	64	160
G.726	20	5	32	80
G.728	10	5	16	60
G.729A	10	10	8	20
G.723.1	20	30	5.3	20

Link Adaptation mechanism (multi-rate channel) is one of the most important factors affect the voice call, several studies have been shown that growth of available bandwidth as a consequence, increases QoS when using adaptive codec selection method. Adjustment of VoIP codec rate based on network conditions, maintains an efficient utilization of the available resources. Sfairopoulou et al. [5] proposes a simple algorithm for dynamic selection of voice codec, depending on network conditions during the on-going voice session.

VoIP over MANETs (VoMAN) have been studied in [6],[7]and [8], these studies show how hops number affect significantly the QoS. The authors in [5] 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 [8].

Designing a routing protocol able to support VoIP calls over a mobile ad hoc network is not straightforward. In one side, VoIP calls are very sensitive to packet losses which are highly related to the system saturation. In other side, VoIP application has a very stringent delay constraint, and should be bounded on transmission delay to assure correct communication [9]. The route re-discovery process in the MANet routing protocol may lead to packet losses and/or delays that affect the quality of ongoing VoIP calls. Knowing that, audio codecs add additional delay to the total network delay.

Certainly, MANets applications are concerned by transmitting voice between network entities. One of the most important applications that need voice communication is the the tele-emergency system, which is in most application cases, very remote regions and disaster struck areas lack telecommunication infrastructure. Therefore, the investigation of mobile ad hoc networking for voice communication is needed. However, Multi-hop voice delivery through MANet is challenging, since it must provide QoS provisioning by efficiently handling constraints of node speeds, unreliable connectivity, rapid topology changes and a

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fragmented network. Primary challenge in designing VoMAN is to provide good delay performance, and handling packet losses.

Knowing that voice quality is mainly influenced by the choice of codecs, we analyzed the performance evaluation of ITU-T audio codec standards in context of MANet voice call. In order to assess these codecs, a series of simulations have been carried out based on real data of an urban environment. The next section describes our measurement methodology.

3. MEASUREMENT METHODOLOGY

Testing the impact of MANet applications before their deployment is an important issue. However, measurement of an actual MANet is expensive and infeasible. Therefore, simulation seems to be the most feasible solution. For this purpose, the discrete event network simulator ns-2 [10] was used combined with a road traffic generator, aiming at a significantly higher level of simulation accuracy. Studies in wireless multi-hop networks highly depend on the scenario considered. This section proposes target scenarios for the study.

The considered scenarios focus on the unicast transmission of voice signals between nodes moving at rate of 0-50 km/h during 200s. In order to analyze how various conditions affect the quality of the voice, two scenarios are considered (details are summarized in Table 2). Each scenario defines a network area sizes, which is simulated with varying conditions: traffic densities and voice calls.

TABLE II. MANET SIMULATION SCENARIOS

	Scenario 1			*Scenario 2*		
Area size	1000m²			2000m²		
Nodes	10	30	50	40	60	80
Voice calls	5	10	20	15	20	30

Reflecting the network interactions in a trustworthy manner is an important issue in simulation. The ns-2 simulator has been widely used for this purpose in MANET evaluation.

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 [11] 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. Further experiments have been tuned OLSR parameters by varying HELLO interval.

Finally, at the application level we used VoIP communication generated via *ns2voip++* [12] module, which is an extensible VoIP application implemented in ns-2. The number of voice call is dependent of simulation scenario and are performed during different moments of the simulation time with duration of 60s. Each VoIP source is configured to draw the duration of the talk-spurt and silence periods from Weibull distribution. Additionally, scenarios have been experimented varying the voice codecs which are previously described in Table 1. Note that we have integrated two other ITU-T codecs (G.726 and G.728) to ns2voip++ framework. Table 3 summarizes some important features of the network used in our VoVAN simulations.

TABLE III. SIMULATION PARAMETERS

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 (Table 1)
Call duration	60s
Simulation time	200s

A. Evaluation metrics:

With the purpose of evaluating the performance of VoVAN system, five metrics have been analyzed. These metrics are associated with user and network level that have a significant influence on perceived speech quality.

Delay: The network delay can be calculated by averaging the End-to-End (E2E) delay which is the time taken for a packet to be successfully delivered from the source to the application layer of the destination, including processing, queuing and propagation delay. According to ITU Recommendation G.114 [13], to achieve good transmission quality, a network delay of no more than 150 ms is required. If the delay exceeds 300 ms, the quality of the VoIP stream is significantly degraded.

Packet loss: For network performance metric, packet loss is measured as the percent of packets dropped at the receiver prior to data stream playback. A loss of 5% or more is usually

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noticeable. Though VoIP applications tolerate packet loss up to 10%, a packet loss of 1% still affects the quality of the VoIP stream. VoIP is a real-time audio service that uses UDP. Because of an unreliable protocol, the recovery of lost packets is not possible, so the codecs must be able to handle some packet loss.

Mean Opinion Score (MOS): used to express the human opinion about QoS. ITU-T P800 [14] defines MOS as a subjective metric which estimates the user satisfaction by means of a score which varies from 1.0 (poor) to 5.0(best). However, A MOS below 3.6 results in many users who are not satisfied with the call quality. In order to assess perceived voice quality, we adopted objective method (PESQ) [15] to obtain MOS score, by converting R-Factor scale obtained by the following expression:

$$R = R0 - Is - Id - Ie + A$$

Where: R represents the result voice quality (from 0 to 100), R0 refers to basic signal-to-noise ratio, Is characterizes the simultaneous impairment factor such as too load speech level, Id represents mouth-to-air delay, Ie is the equipment impairment factor (e.g. codecs, packet losses and jitter), and A is the advantage of access.

Throughput: considered as the most straightforward metric for the MANET routing protocols. It is computed as the amount of data transferred (in bit) divided by the simulated data transfer time.

Normalized routing Load (NRL): 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.

4. RESULTS AND DISCUSSION

The empirical results presented in this paper may provide a strong reference to deployment of VoIP services through MANet. The overall system performance was tested using the average results obtained from 30×10 MANet scenarios defined in the specification presented above. The analysis focus on the results considering two urban scenarios sizes. This section shows and discusses our simulation results exhibiting the impacts of audio codec on QoS measured in terms of end-to-end delay, packet loss and MOS. In turn, the impact of routing parameters has been considered, in order to make a trade-off with voices codecs.

A. Impact of audio codec on QoS

In Figure 4, we show the average end-to-end network delay obtained using different audio codecs. Looking at this figure it can be seen that in small-scale scenario, all codecs does not exceed the threshold recommended by ITU-T (150 ms) [20]. This is due to network load (relatively low). While in large-scale 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, 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.

Examining the packet loss indicator for all codecs (Figure 5), we can check that it exceed 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 (based on table 2), 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 node spends some time to locate a route to destination, the VoIP source continues to produce packets. When 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 small scenario size because the traffic is slower (34 packet/sec) compared to other codecs. Generally, losses increase in large scenario area because the number of connections increases.

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.

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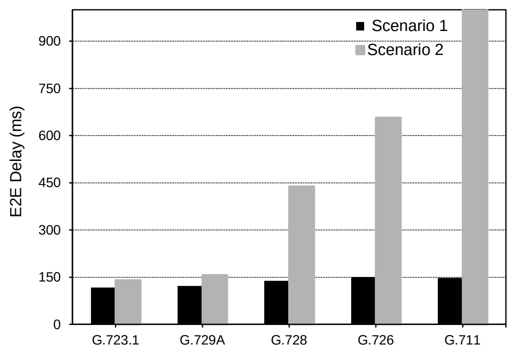


Figure 2. End-To-End Delay For Different Audio Codecs

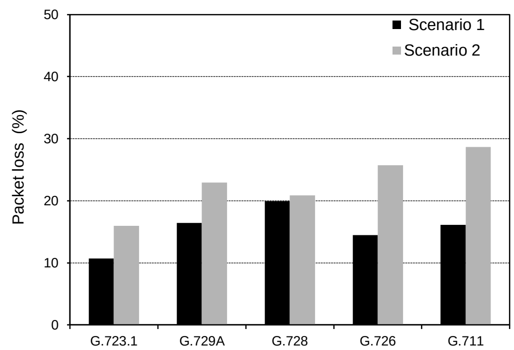


Figure 3. Packet Loss For Different Audio Codecs

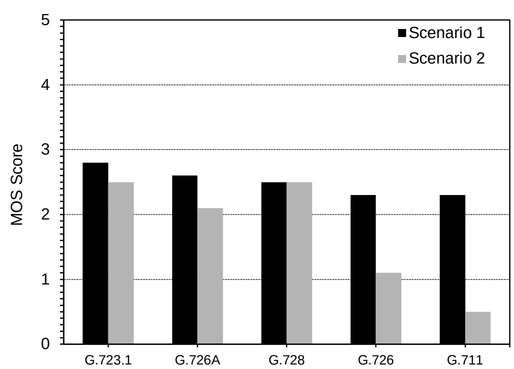


Figure 4. MOS Score For Different Audio Codecs

B. *Impact of tuning OLSR routing parameters*

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. signaling.

In order to evaluate the impact of OLSR soft state signaling, in this section, we measure VoIP flow throughput, control traffic overhead, packet loss and E2E delay for each codec, with respect to different HELLO message intervals. Note that, we have done some modification on OLSR implementation [16] to support decimal value for HELLO intervals.

Figure 5 plots average throughput that gives idea on bandwidth which needs to be occupied by each codec. We observe that increasing/decreasing neighbor 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 6 shown 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 behavior of OLSR is changed depending on the codec used.

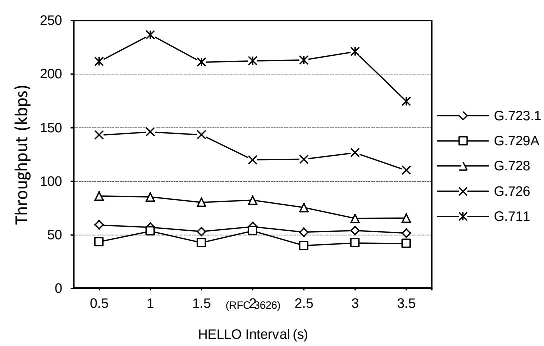
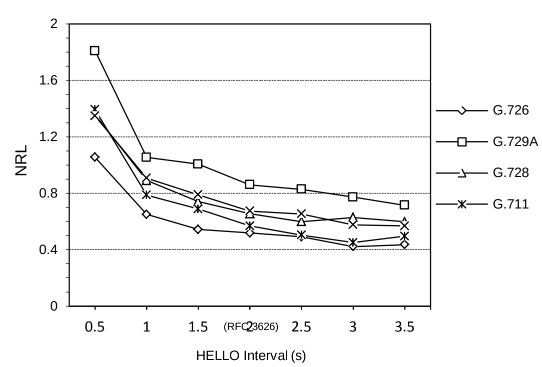


Figure 5. *Throughput Vs. HELLO Intervals*



Fifure. 6 Normalized Routing Load Vs. HELLO Intervals

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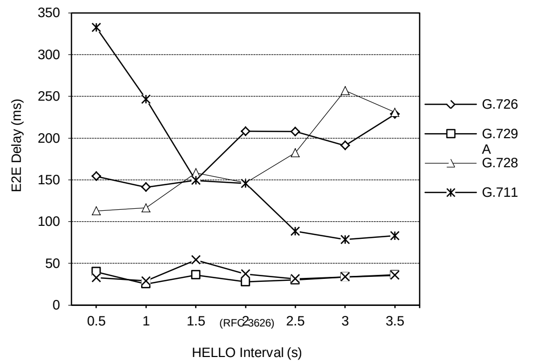

Figure 7. E2E Delay Vs. HELLO Intervals

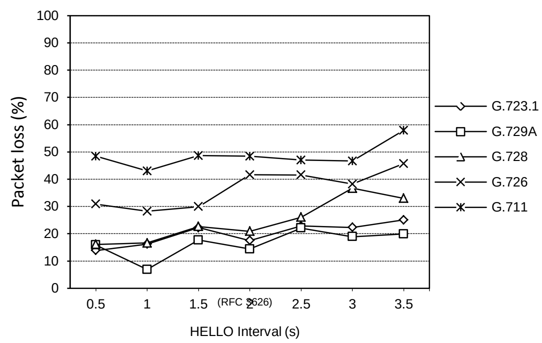

Figure 8. Packet Loss Vs. HELLO Intervals

From the point of view of E2E delay, as shown in figure 7, HELLO interval has not affect delay for codec with low bit-rate. On the other side (codec with high bit-rate) average delay is changed significantly with increase/decrease of neighbor 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.

In term of packet loss, values depicted in Figure 8 shown 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, give packets the ability to reach their destinations. Nevertheless, codecs must be more able to handle losses.

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

5. RELATED WORK

This section overviews works on voice over IP in wireless local area networks (WLAN) and mobile ad hoc networks.

A number of works consider voice capacity of WLANs. In [17], 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. Adaptive modulation and adaptive compression have been applied separately in VoIP-based wireless and wired networks. Sfairopoulou et al. [5] propose a codec adaptation algorithm based on perceived speech quality, which allows a cell-wide optimization of network resources and voice quality on multi-rate WLANs; the results have shown how adaptation increases the performance of the WLAN.

Several works have been focused on voice support over ad hoc networks. In [18], VoIP capacity has been analyzed for different ITU G.711, G.729, and G.723 voice codecs using IEEE 802.11b DCF access scheme at 11 Mbps, results shows voice capacities similar to our experimental results. The small difference between our results may be due to the use of a packet-loss rate below 1% compared to our 10%.

MANet performance evaluation is being extensively studied by several researchers. Very few papers, on the other hand, have addressed the problem of suitability of mobile ad hoc networks for real-time traffic transmission. In [19], 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 which nearly confirms our results.

To the best of our knowledge, this is the first work that addresses the impact of tuning routing parameters on voice codecs performance. However, according to results presented by several works, OLSR has a range of improvement by changing the configuration parameters. In [22], authors investigate the different impacts of tuning

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refresh interval timers on OLSR protocol performance; as result, authors found that reducing refresh intervals could improve performance. Gomez et al. [21] define Route Change Latency (RCL) metric and experiment with a set of OLSR settings in real ad hoc network. Results showing that end-to-end connectivity can be enhanced using different parameter settings from the default ones.

6. CONCLUSION & FUTURE WORKS

Real-time voice transmission over mobile ad hoc network is very much demanding and necessary, especially in emergency scenarios. This work aims at providing meaningful results to guide the design of efficient strategies and protocols to support VoIP communications over MANets.

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 successfully deal with VoIP applications requirement, especially over large-scale MANet environment.

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 codec/routing adaptation mechanism may be an emerging solution for maximizing voice quality and provide suitable QoS for VoMAN.

As a matter of future works, we intend to design a QoS management system for VoMAN based on policies and cross layer architecture. In addition, we are extending planning outdoor test to validate the simulation results.

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