Analysing End-to-End Packet Delay and Loss in mobile ad hoc networks for interactive audio applications

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1 Introduction

Interactive audio applications such as audio conferencing and telephony require high constraints on delay, jitter and loss. The packets of these applications must be received without significant loss, with low delay and jitter. When packet loss rate exceeds 10 % and one way delay exceeds 150 ms, speech quality can be quite poor. Human conversation tolerates a maximum end-to-end delay of between 150 and 300 milliseconds [IT01]. In addition, these packets must have a small delay variation to maintain constant rate for successive audio packets at the destination.

In ad hoc networks [adh00], many factors such load traffic, codec bit rate, routing protocol configuration and mobility speed have an impact on packet loss rate, average delays, and increased jitter which degrade the quality of the received audio signal. In this paper, we analysed how these factors influence packet delay and loss behaviour in ad hoc networks. A best knowledge of this behaviour is important to develop more effective mechanisms for dynamic adjustment of playout delays or throughput, in order to improve the perceptual quality of audio applications running in such environment. Our study is done by simulations, where we distinguish two operating phases of an audio flow, normal phase (without link breakage) and reconfiguration phase (at the presence of mobility).

The rest of this paper is organised as follows: Section 2 describes the environment and parameters of simulations; Section 3 presents the results of the simulations and analyses them. Section 4 conclude this paper.

2 Simulation model and methodology

Our simulations are performed using the Ns2 (Network Simulator) [Gre01][ISI01].

2.1 Simulation environment

Our simulation modeled a network of 50 mobile hosts placed randomly within a 1000 meter x 1000 meter area. Radio propagation range for each node was 250 meters and channel capacity was 11 Mb/s. The IEEE 802.11 MAC protocol [IEE99] is used as the MAC layer in our simulations. The specific access scheme is Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA) with acknowledgements.

2.2 Traffic pattern

A traffic generator was developed to simulate constant bit rate (CBR) and audio sources.

2.2.1 Audio traffic

An audio traffic is generated between a source and a destination randomly selected. An audio flow is typically divided into "talkspurt" (periods of audio activity) and "silence periods" (periods of audio inactivity, during which no audio packets are generated). We consider an average talkspurt of 30.83% and average silence period of 61.47% as recommended by the ITU-T specification for conversational speech [IT01]. The alternating periods of activity and silence are exponentially distributed with average duration of 1.004 s and 1.587 s, respectively. During activity periods, 320 bytes voice packets were generated periodically according to the used audio codec. In our simulations, we use 2 codecs with different bit rates (PCM[1] at 64 kbps and ADPCM3[2] at 24 kbps) to generate audio traffic.

1. Pulse Codec Modulation- See Recommendation G.711 in [IT01]
2. Adaptive Differential PCM 3 - See Recommendation G.721 in [IT01]

We examine the behaviour of delay and loss of the audio flow when we change the codec bit rate. It is essential to compare audio performance with different sending bit rate, particularly in constrained bandwidth networks such as ad hoc network.

2.2.2 CBR traffic

Constant Bit Rate (CBR) data sessions with randomly selected sources and destinations were simulated. Each source transmits data packets at rate 10 packets/sec. The size of data is 512 bytes. We vary the traffic load by changing the number of data sessions and examine its effect on delay and loss of the audio flow.

2.3 Routing protocol

The Ad hoc On-Demand Distance Vector routing (AODV) and Optimised Link State Routing (OLSR) protocols are used as representatives of reactive and proactive routing protocols, respectively [EMR00][JMQ00]. The table below shows the AODV and OLSR parameter values used in our simulations. We examine the effect of these two routing approaches on delay and loss of the audio flow.

Parameter for AODV	**Values (seconds)**	**Parameter for OLSR**	**Values (seconds)**
HELLO interval	1.0	HELLO interval	2.0
Route reply wait time	1.0	Neighbour entry time-out	6.0
Maximum route request time-out	10.0	Topology entry time-out	16.0
Active route time-out	50.0	Topology control interval	5.0

2.4 Mobility

During an active period of the audio flow, we distinguish two phases:

– Normal phase: during which the flow reaches the destination from the source on the same route (without link breakage). This phase is simulated in a network without mobility to examine the behaviour of delay and loss.

– Reconfiguration phase: during which nodes move on the current route and the audio flow transfer is interrupted for delay required from a routing protocol to establish a new route towards the destination. This phase is simulated in a network with mobility. We vary the node mobility speed from 2 m/s to 20 m/s to examine the behaviour of delay and loss during this phase.

2.5 Simulation and performance parameters

Multiple runs with different seed numbers were conducted for scenario and collected data was averaged over those runs. Each simulation executed for 300 seconds. The averaged performance parameters are:

1. End-to-end delay is the total time between a packet's transmission and it's playout at the receiver.
2. Loss percentage is the percentage of lost packets due to congestion or overloading in the network.
3. Loss late percentage is the percentage of lost packets due to late arrivals. When end-to-end delay exceeds 300 ms, the packet is considered lost for interactive audio application.
4. Total loss percentage is the percentage of lost packets including late arrivals and network losses.

3 Results and analysis

This section presents results and analysis of our simulations.

3.1 Normal phase (network without mobility)

During a normal phase, the factors which influence delays and loss are essentially network load traffic and codec bit rate.

3.1.1 Effect of network traffic load

Figures 2, 3 and 4 show the variation of delay, network loss and late loss percentage, respectively, for the audio flow. Each figure presents two graphs for AODV and OLSR routing protocols. In light load (ie., less than 10 sessions), delays for AODV and OLSR are nearly equal and increase when increasing the number of sessions. Delays for reactive scheme become longer than for proactive scheme. When nodes are not mobile, routing table for available destinations is already known for OLSR, while, AODV needs delay to discover route for required destination just before sending. So, loss due to late arrivals are more important with AODV compared to OLSR. We can remark that AODV outperforms OLSR when comparing packet reachability as shown in figure 4. Packet loss is more probable in OLSR than in AODV due to the fact that AODV is based on broadcasting in case of route repairing. In some cases, to route packets, broadcasting is more efficient than using a fully distributed protocol such OLSR. Figure 5 shows the total loss percentage for OLSR and AODV. So, the total loss percentage for AODV increases rapidly when increasing load traffic (ie., more than 14 sessions) and is clearly larger than for OLSR.

3.1.2 Effect of codec bit rate

Figures [6, 7, 8] and [10, 11, 12] present the variation of delay, network loss and late arrivals for AODV and OLSR, respectively. Each figure presents two graphs for two codec bit rate. These figures reveal that for both routing protocols, when using ADPCM3 codec, delays, network loss and late arrivals become less longer than when using PCM codec. Naturally, when decreasing sending bit rate, congestion and queueing in the network will be less probable. When computing total loss percentage including network loss and late arrivals, plotted in figure 9 for AODV and in figure 13 for OLSR, we conclude that with ADPCM3 codec, loss percentages for both routing scheme are clearly less than with PCM codec. In addition, for both routing protocols loss percentage is around 10 %. When using a codec with high bit rate, loss percentage can exceed 15 % particularly for OLSR in high load traffic (see figures 12). Also, note that OLSR outperforms AODV with ADPCM codec when comparing delays and late arrivals for the same reasons described in the previous subsection.

Our conclusion is that when using a codec with low bit rate, we lead to better audio performance with both routing protocols, particularly with OLSR routing. OLSR outperforms AODV when comparing total loss due to network loss and late arrivals, when increasing load traffic (ie., more than 10 or 12 sessions in a wide range of scenarios).

3.2 Reconfiguration phase (network with mobility)

A reconfiguration phase constitutes a sudden, large increase in end-to-end network delays, followed by series of simultaneous packets arriving with high end-to-end delays variation (jitter). A reconfiguration phase can be characterised by its duration, delay and jitter amplitude. For this phase, we study the delay and loss behaviour during all the audio session and during the phase its self. A reconfiguration phase is illustrated in figure 1.

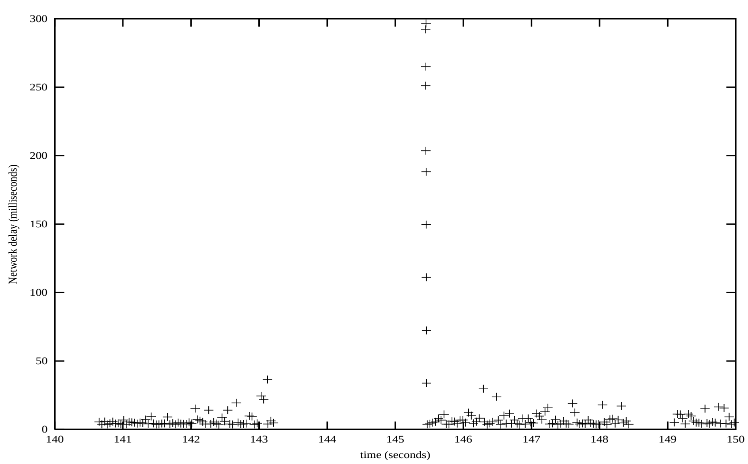

FIG. 1 – *Example of a reconfiguration phase (End-to-end delays as function of packet arrival time)*

3.2.1 Analysis during a reconfiguration phase

We analyse effect of mobility, load traffic and codec bit rate on delay and loss. This work is in progress.

3.2.2 Analysis during all the audio session

The factors which influence delays and loss are essentially node mobility speed and codec bit rate.

1. Effect of mobility speed: Observing figure 14, we remark that OLSR leads to longer delays, consequently to higher loss late percentage, than AODV as speed mobility increases. AODV protocol reacts faster than OLSR to changes in network topology. OLSR will have to detect broken links, select MPRs, and diffuse the new

topology information into the network, which will take more delay than for AODV. AODV will only have to detect that the link has broken before performing a route repair or a route request. Thus, OLSR requires more delay to establish a new route towards the destination which in turn can be the reason of increased lost packet for OLSR compared to AODV (see figures 16 and 15). Consequently, It results that the total loss percentage for OLSR is more important than for AODV (see figure 17) . we note that the impact of node mobility speed on delays, loss and late arrivals become more significant when increasing load traffic.

2. Effect of codec bit rate: Figures 18, 19 and 20 show the variation of delay, network loss and late arrivals for AODV with two codec bit rates. In network with medium speed (2 to 8 meters/second), network and late arrivals losses increase as mobility speed increases. The mobility speed has not a significant impact on AODV when it become greater than 8 meters/second. Network and late arrivals losses with ADPCM3 codec are nearly constant and lower than with PCM codec as shown in figures 19 and 20. We remark in figure 18 that delays for AODV increase as mobility speed increases whereas loss due to late arrivals does not continuously increases as shown in figure 19. Observing simulations in more details, we constant that some packets experience very high delay to reach destinations, which influence the averaged delays values. This is not the case of OLSR. Certainly, when using ADPCM3 codec with OLSR routing, delays, network and late arrivals losses become lower than with PCM codec but increase when increasing mobility speed (see figures 22, 23, 24). So, the total loss percentage for both routing protocols become lower when using less sending bit rate codec and it is lower for reactive scheme compared to proactive scheme in mobile environment (see figures 21 and 25).

4 Conclusion

In this paper, we have focused on the effects of traffic load, codec bit rate and mobility on delay and loss in mobile ad hoc networks for interactive audio flow. Through analysis of different scenarios, we conclude that in network with no mobility, the performance of interactive audio flow is better when using OLSR routing, while in network with mobility, AODV is better. Only in network with very light load and low mobility, OLSR performs just as good as AODV. In addition, for both routing protocols, when using sending bit rate at 24 kbps, loss percentage is around 10 %. This can satisfy audio interactive constraints. It is more efficient to use a codec at less bit rate to obtain tolerable loss percentage. When using a codec with high bit rate, loss percentage can exceed 15 % and can seriously degrades the audio quality. It seems interesting to adjust dynamically the codec bit rate according to a feedback report, where it is sufficient to include total loss informations.

Note that these results are consistent with claims in [JMQ00] that say that OLSR performs just as good as AODV, but has important an substantial advantages in particular scenarios such as networks with highly sporadic traffic and high density, while AODV performs better than OLSR in networks with static traffic. The last case corresponds to parameter environment in our simulations, where traffic is static during the same simulation session. In addition to the claimed conclusions, we say that in no mobile networks with static traffic, OLSR performs better than AODV.

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5 Normal phase

5.1 Effect of load traffic for AODV and OLSR (with PCM Codec: 64 Kbps)

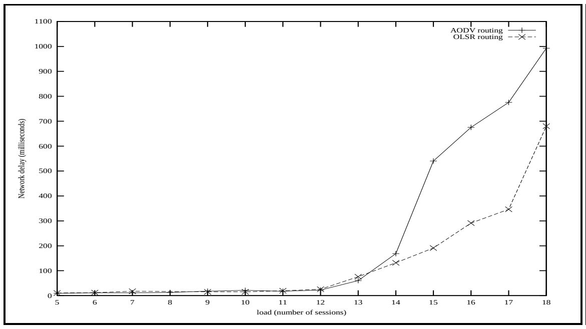


FIG. 2 – ***Averaged end-to-end delay as a function of load traffic (AODV and OLSR)***

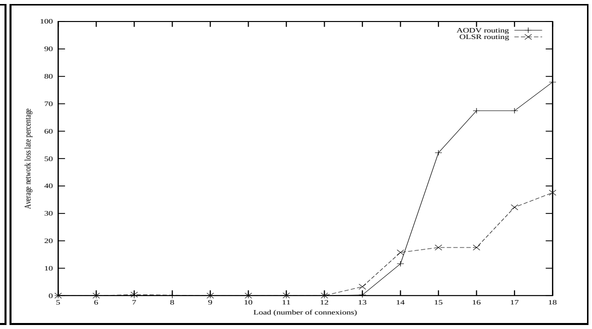


FIG. 3 – ***Averaged loss late percentage as a function of load traffic (AODV and OLSR)***

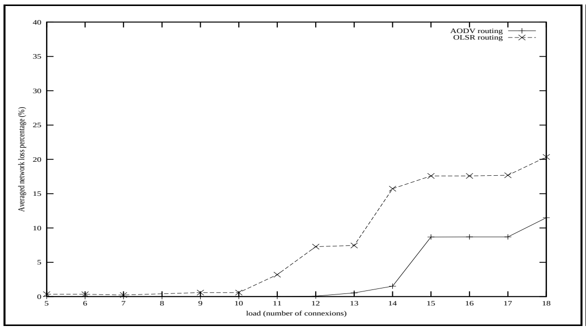


FIG. 4 – ***Averaged network loss percentage as a function of load traffic (AODV and OLSR)***

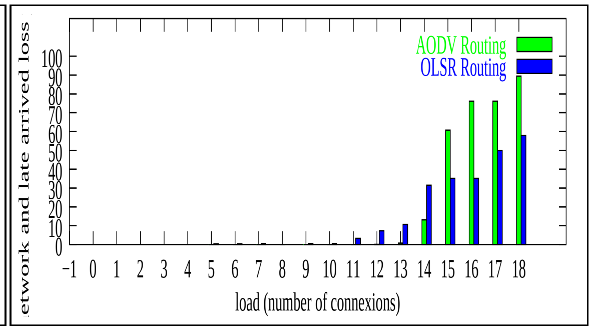


FIG. 5 – ***Averaged total loss percentage as a function of load traffic (AODV and OLSR)***

5.2 Effect of bit rate codec for AODV (PCM Codec: 64 Kbps and ADPCM3 Codec: 24 kbps)

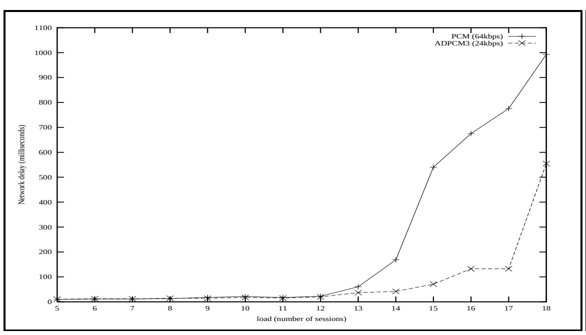


FIG. 6 – ***Averaged end-to-end delay as a function of load traffic (AODV)***

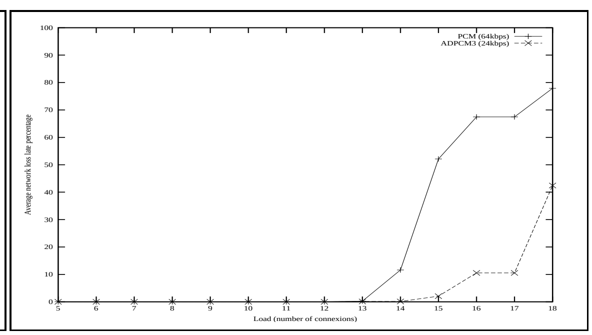
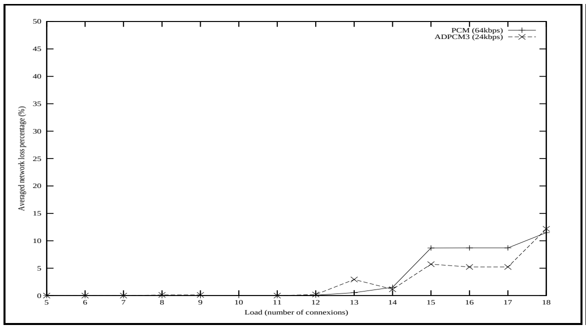


FIG. 7 – ***Averaged loss late percentage as a function of load traffic (AODV)***



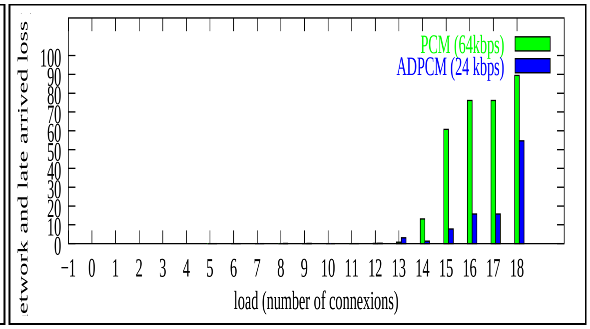
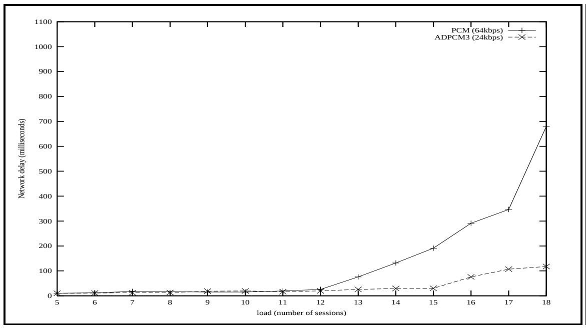


FIG. 8 – ***Averaged network loss percentage as a function of load traffic (AODV)***

FIG. 9 – ***Averaged network and late arrivals loss percentage as a function of load traffic (AODV)***

5.3 Effect of bit rate codec for OLSR (PCM Codec: 64 Kbps and ADPCM3: 24 kbps)



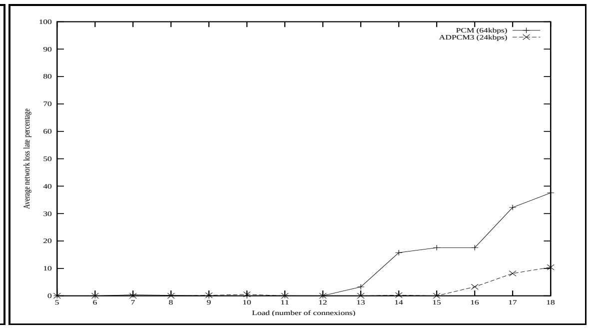
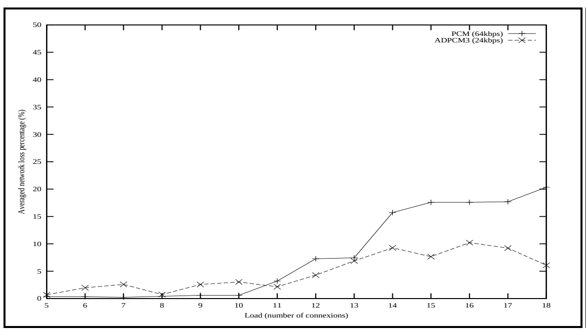


FIG. 10 – ***Averaged end-to-end delay as a function of load traffic (OLSR)***

FIG. 11 – ***Averaged loss late percentage delay as a function of load traffic (OLSR)***



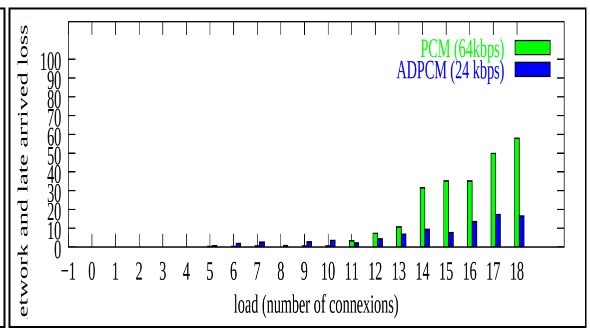


FIG. 12 – ***Averaged network loss percentage as a function of load traffic (OLSR)***

FIG. 13 – ***Averaged network and late arrivals loss percentage as a function of load traffic(OLSR)***

6 Network with mobility

6.1 Effect of mobility speed for AODV and OLSR with PCM Codec at 64 kbps

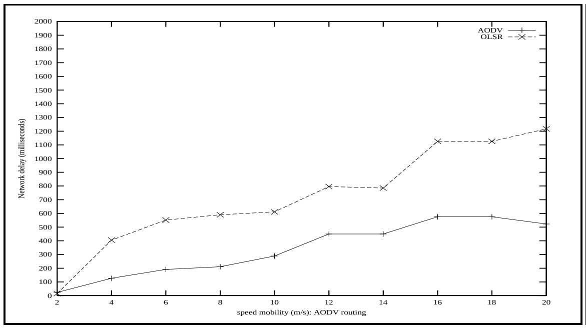


FIG. 14 – ***Averaged end-to-end delay as a function of mobility speed: AODV and OLSR***

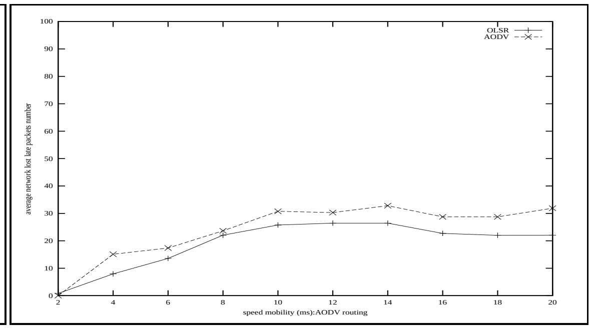


FIG. 15 – ***Averaged Averaged loss late percentage as a function of mobility speed: AODV and OLSR***

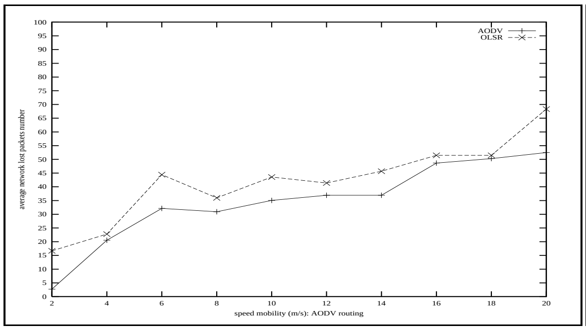


FIG. 16 – ***Averaged network loss percentage as a function of mobility speed: AODV and OLSR***

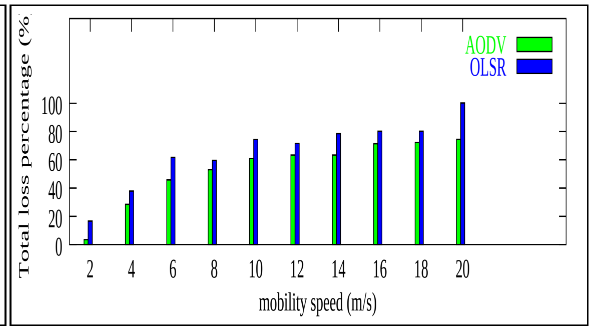


FIG. 17 – ***Averaged total loss percentage as a function of mobility speed: AODV and OLSR***

6.2 Effect of codec bit rate for AODV (PCM Codec: 64 Kbps and ADPCM3: 24 kbps)

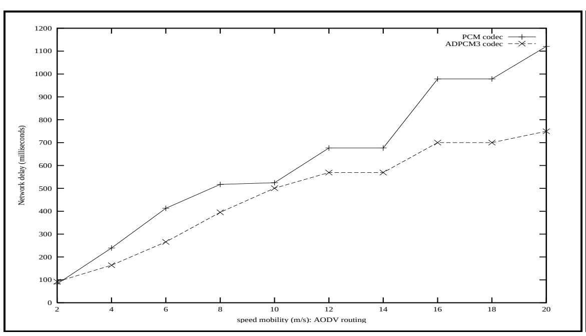


FIG. 18 – ***Averaged end-to-end delay as a function of mobility speed: AODV routing***

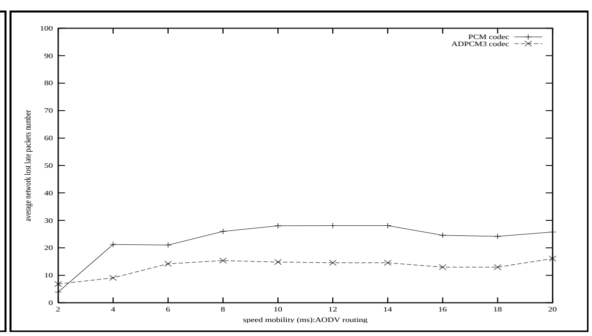


FIG. 19 – ***Averaged Averaged loss late percentage as a function of mobility speed: AODV routing***

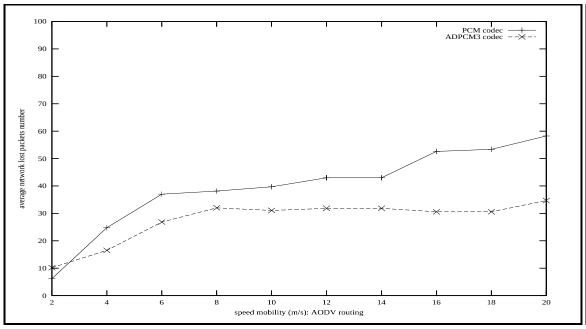


FIG. 20 – ***Averaged network loss percentage as a function of mobility speed: AODV routing***

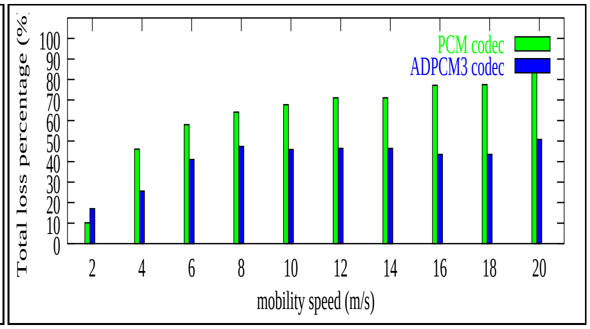


FIG. 21 – ***Averaged total loss percentage as a function of mobility speed: AODV routing***

6.3 Effect of codec bit rate for OLSR (PCM Codec: 64 Kbps and ADPCM3: 24 kbps)

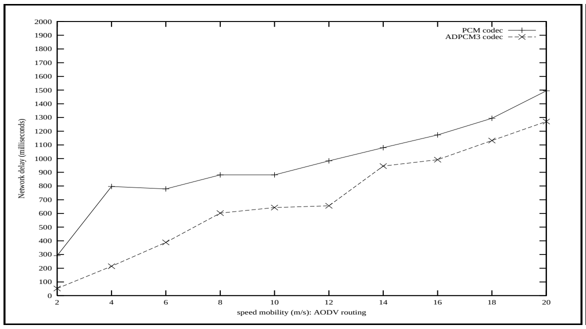


FIG. 22 – ***Averaged end-to-end delay as a function of mobility speed: OLSR routing***

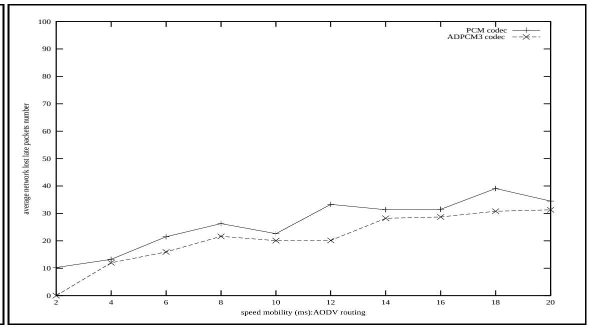


FIG. 23 – ***Averaged Averaged loss late percentage as a function of mobility speed: OLSR routing***

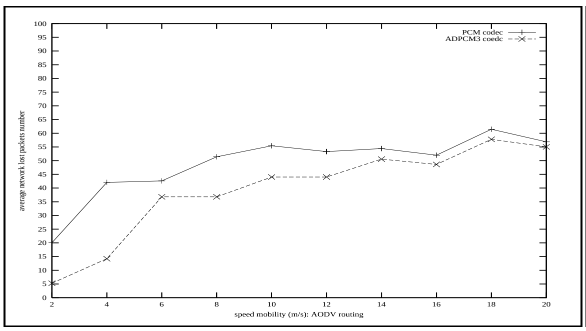


FIG. 24 – ***Averaged network loss percentage as a function of mobility speed: OLSR routing***

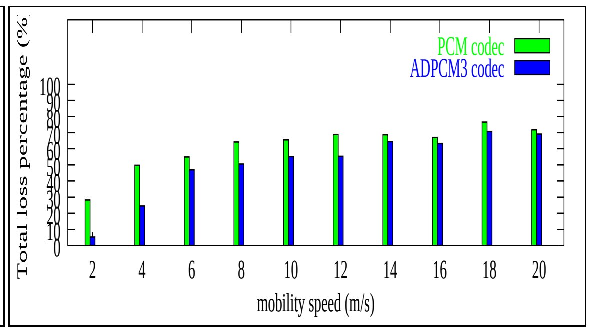


FIG. 25 – ***Averaged total loss percentage as a function of mobility speed: OLSR routing***