A Comparative Study of Voice over Wireless Networks Using NS-2 Simulation with an Integrated Error Model

Bilal Munir, Naveen K Chilamkurti, Ben Soh
Dept of Comp Science and Comp Engineering
La Trobe University
Bundoora, Melbourne, Australia-3086

Abstract - **Wireless communication is the fastest growing field and with the emergence of IEEE 802.11 based devices, wireless access is becoming more popular. Many multimedia applications for IP networks have been developed and thus the demand for quality of service (QoS) has increased. In this paper our primary objective is to evaluate 802.11e EDCF framework for video, voice and background traffic all at the same time. Our assessment is based on an error model called E-model, MOS for VoIP and PSNR for video. We also studied the effects of Random Uniform error model on various types of traffic. As expected, wireless networks are more prone to errors than wired networks.**

Keywords: Voice over Wireless, 802.11, 802.11e, NS-2.

1. INTRODUCTION

The emerging IEEE 802.11 technology has been growing tremendously in the last few years. With the improvement of the technology, the need for providing better service is becoming more and more vital for manufacturers. The legacy 802.11 networks are of best effort and therefore do not support Quality of Service (QoS) for time critical applications [1, 2].

In wireless networks, IEEE 802.11e has replaced best effort services with improved services which provide QoS. IEEE 802.11e introduces two additional MAC modes: the Enhanced Distributed Coordination Function (EDCF) and the Hybrid Coordination Function (HCF). HCF contains two medium access mechanisms: HCF controlled channel access (HCCA), and enhanced distributed contention channel access (EDCA). HCCA mechanism provides parameterized QoS service for contention free transfer. EDCA mechanism provides prioritized QoS services for contention-based transfer. At present, access point only supports EDCA function due to complexity of HCCA implementation. Many researchers have used different experimental setups to evaluate 802.11e QoS most of which were based on network level analysis [3-7], whereas some other researchers have used similar experimental arrangements with different network topologies.

The motivation of this paper is to present an 802.11a and 802.11e comparison framework to evaluate VoIP, video and background traffic at the same time using Network Simulator (NS-2) [8] to simulate IEEE 802.11e

EDCF mode. We also study the effects of errors with changing background traffic when an error model is introduced. Our EDCF model does not include contention free bursting (CFB). We apply our simulation model to evaluate the VoIP quality of calls using EDCF model and video quality using PSNR in the presence of CBR background traffic.

In our research, for video quality we have used Peak Signal to Noise ratio (PSNR) and Perceptual Evaluation of Speech Quality (PESQ) R-Factor for voice quality [9]. We used H.264 codec to encode recorded videos because of its better compression capabilities and used ITU G.729 codec for voice encoding.

Section 2 of this paper discusses metrics for Video quality and VoIP quality. This section also discusses the flow of video and voice simulation in NS-2. Section 3 discusses simulation and evaluation environment. In section 4 we discuss the results and section 5 presents conclusions and future directions of the work.

2. TECHNICAL BACKGROUND

In 802.11 based wireless networks, two types of network topologies are used: infrastructure mode and ad-hoc mode. In infrastructure topology, all wireless nodes communicate through the central access point. In ad-hoc, wireless nodes can talk to another node by constructing a path through other nodes without access point. This paper discusses the transmission in infrastructure topology.

A. VoIP Quality

The quality of service of IP telephony is a subjective quality which corresponds to the user perception of quality of transmitted speech. The VoIP quality is dependent on many parameters like delay, jitter, packet loss, media bandwidth, and echo suppression. The Mean Opinion Score (MOS) test is commonly used for speech quality rating, but it is time-consuming and expensive. Several objective MOS measures are in use, such as Perceptual Analysis Measurement System (PAMS) and Perceptual Evaluation of Speech Quality (PESQ). They measure the audible distortions based on the perceptual domain representation of two signals, a reference signal and a degraded signal which is the output of the system under test [10]. The MOS value ranges from 1(bad) to 5 (excellent) speech qualities. Another approach is ITU-T G.107, the E-model [11], a computational model which combines all parameters to give a single total value called the R factor, which ranges from 0(bad) , >70(toll quality) to 100(excellent). To calculate R factor, E-model considers the speech quality, the mouth–to-ear delay, echo and loudness.

B. Video Quality

To measure the video quality Peak Signal to Noise Ratio (PSNR) is used. PSNR is measure of distortion in transmitted

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signal due to compression or transmission. Two mathematical formulae are used to measure PSNR.

$$\text{PSNR} = 10\log\left[\frac{Q^2}{\text{MSE}}\right] \quad (1)$$

$$\text{MSE} = \frac{\sum_{i=1}^{N}\sum_{j=1}^{N}\left[f(i,j) - \hat{f}(i,j)\right]^2}{N^2} \quad (2)$$

In the formula 1, Q is maximum sample size. If the frame size is NxN then Mean Square Error (MSE) calculates the difference of each pixel between original image and distorted image. The smaller MSE is, the higher PSNR achieves and lower distortion degree [12, 13].

B. *Wireless Error Models*

Two types of error models have been widely used in wireless networks, Random Uniform error model and Gilbert-Elliot (GE) error model. The error pattern generated by Random Uniform is uniformly distributed. Usually wireless errors are burst pattern and GE is a good approximation of such errors. In this paper we have used Random Uniform Error model for simulation.

3. Simulation and Evaluation Environment

The goal for the simulations is to study the extent of the quality of the VOIP and Video traffic decreases with various error rates, while changing the background traffic. Our paper is an extension of the work carried out by C. H. Ke et al [7, 12] with few changes. We used network simulator NS-2 with IEEE 802.11e EDCF.

C. *Audio Simulation*

Figure 1 describes the voice simulation. For simulation purposes we recorded a human voice in .wav format, encoded it first in 16 bit PCM format and then in G.729 format to get a bit file. This bit file is then passed through a packetizer to convert it to PTR/UDP/IP packets.

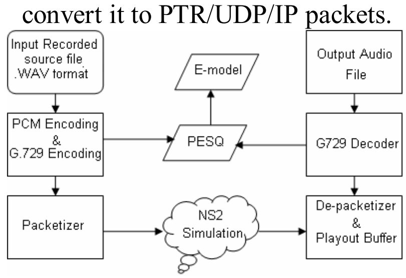

Figure 1: Audio simulation

The sender sends one packet per 20ms, and the packet size is 160 bytes. At the receiver end, the received file is recorded and a distorted bit output trace file with packet id and receiving time is generated. To implement the playout buffer we used Moon's and Ramjee's algorithm [14, 15]. Finally the distorted bit file is decoded by G.729 decoder and PESQ-MOS and R factor is calculated.

D. *Video Simulation*

PSNR is the measure of the Luminous factor (Y) of a video file. For simulation purposes we took a raw video YUV file and encoded it with H.264 encoder to obtain an H.264 bit stream file. This bit stream file is then passed from parser/packetizer to get an input trace file with Packet id, size and sending time information. At the receiver end, another output trace file with similar information is recorded. By comparing original bit stream file and the output trace file we will get lost packets information in a distorted bit stream file. This file is then decoded by decoder to generate a distorted YUV file and could be viewed by a YUV viewer. The raw YUV file and the distorted YUV file are compared for PSNR.

E. *Experimental Setup*

Figure 2a shows our first experimental setup and various parameter settings for this paper. There are 3 sender nodes 1) CBR background traffic generator, 2) VoIP traffic source and 3) H.264 Video source. All 3 links can pass traffic at 10Mbps, 1msec

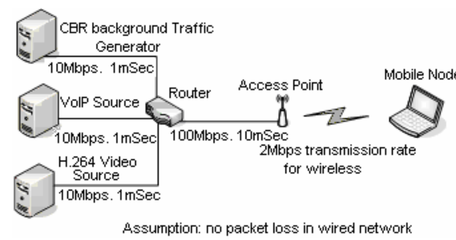

Figure 2a: Simulation setup (one mobile receiver node)

to a router and this router passes the traffic at 100Mbps,10msec to an Access Point which further transmits this traffic to a mobile node at a transmission rate of 2Mbps. Section 3.4 lists all the parameter settings for our experiment.

Figure 2b shows our 2nd experimental setup. All settings are same in this setup except now we have two mobile nodes receiving the traffic from all 3 sources.

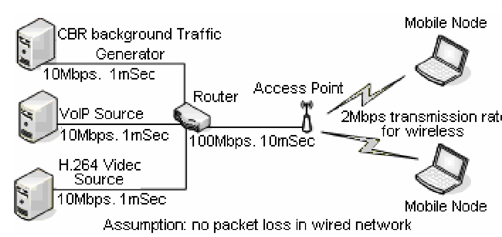

Figure 2b: Simulation setup (2 mobile receiver nodes)

F. *Simulation Parameter settings*

Table 1. EDCF parameter settings

	Voice	*Video*	*Background*
AIFS	2	2	3
CWmin	7	15	31
CWmax	15	31	1023
PF	2	2	2

802.11a parameter settings

CWmin	*CWmax*	*Slot time*	*SIFS*
15	1023	9 micro Sec	16 micro Sec

Voice parameter settings

We recorded our own voices using windows voice recorder and saved as wav files.

We used G.729 codec to encode the wav files and put two frames into same packet and send it as RTP/UDP/IP packet into the simulation environment.

Video parameter settings

We used Qcif format foreman video sequence and encoded 252 frames using H.264 in slice mode. [16, 17]

4. SIMULATION RESULTS

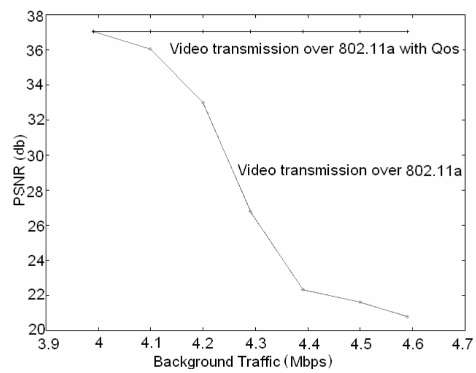

Figure 3: Background traffic vs PSNR without error

Figure 3 shows a graph between Background traffic and PSNR. The encoded PSNR of foreman after H.264 encoding is 37.04. During the encoding process, the video will lose some information and will introduce some distortions. These distortions are due to data compression. The figure shows that with the increase in background traffic the PSNR for 802.11a decreases rapidly and after some time total video traffic is lost, whereas in the case of 802.11e despite of the huge increase of background traffic, PSNR is quite stable, almost same as at the encoder and there is no change at all, which proves that there is no data loss during the transmission.

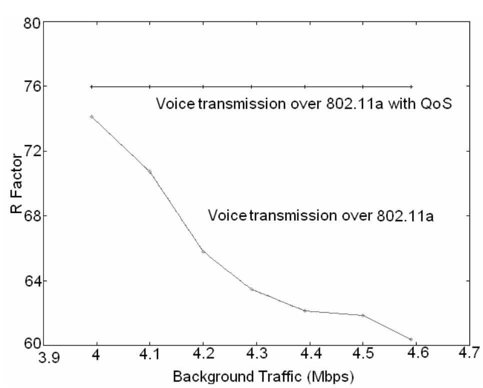

Figure 4: Background traffic vs R Factor without error

Figure 4 shows a graph between Background traffic and Voice quality (R-Factor). When there is low traffic, the difference between 802.11a and 802.11e is not significant but with the increase in background traffic in 802.11a, R-Factor decreases and eventually reaches to a point where almost all voice packets are dropped, whereas, in case of 802.11e, the difference is quite obvious and there is no change in the voice quality and the figure 4 shows a straight line.

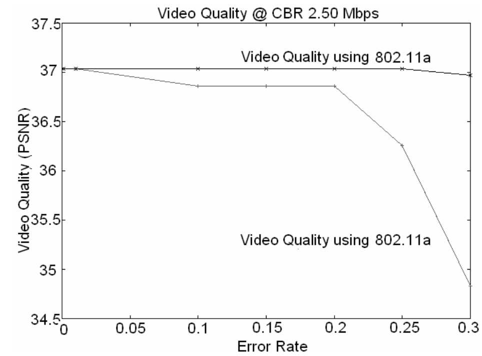

Figure 5: Error Rate Vs Video Quality

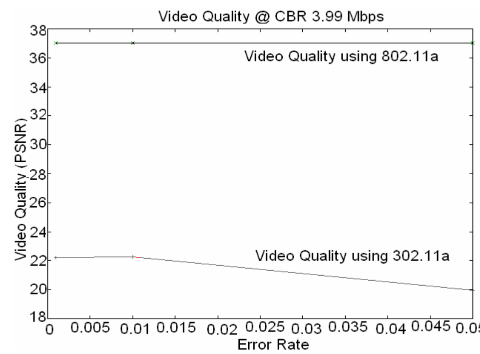

Figure 6: Error Rate Vs Video Quality

Figures 5 and 6 show graphs between Error rate and PSNR with different CBR Background traffic. Figure 5 shows that at 2.5Mbps CBR background traffic and low error rate, both 802.11a and 802.11e are almost same. There is no difference between the two, whereas with the increase in error rate, the difference keeps on increasing and as we pass 20% error rate, there is a rapid drop in PSNR in 802.11a. This drop is because of the increased error rate in transmission. The erroneous packets need to wait until they get retransmitted and have to compete against other traffic follows in 802.11a. In the case of 802.11e, there is no significant change in PSNR. This is because that the video traffic has given a higher priority as compared to background traffic and most of the wireless errors are recovered by retransmission. Figure 6 also verifies the same results at higher CBR traffic.

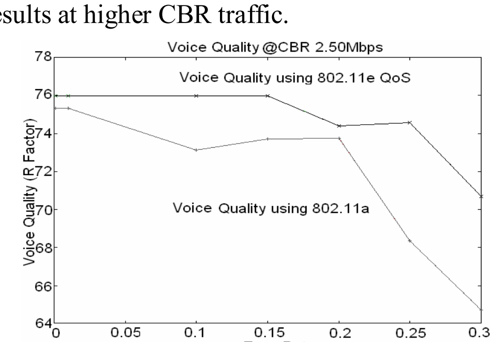

Figure 7: Error Rate Vs Voice Quality

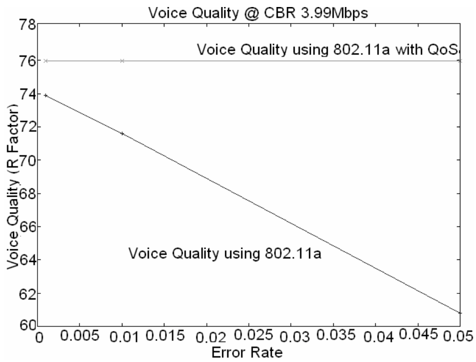

Figure 8: Error Rate Vs Voice Quality

Figures 7 and 8 show graphs between Error rate and Voice Quality at different CBR Background traffic. When there is low traffic, the difference between 802.11a and 802.11e is not significant but with the increase in error rate in 802.11a, R-Factor decreases and at 20% error rate R-Factor drops very rapidly and eventually reaches to a point where almost all voice packets are dropped. This is mainly due to the packet loss because of increased error rate, whereas, in case of 802.11e, the difference is quite obvious and there is also a change in the voice quality around 20% error rate which decreases as the rate increases but the still the voice quality remains better than toll quality (i.e.70%).

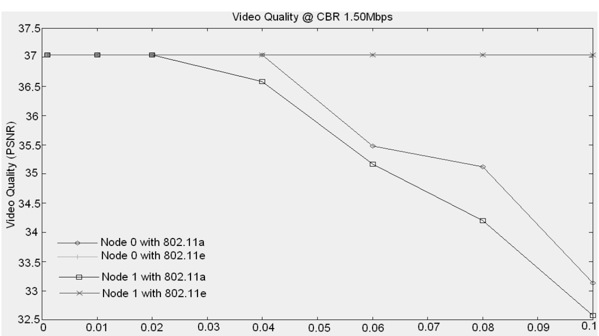

Figure 9: Error Rate Vs Video Quality

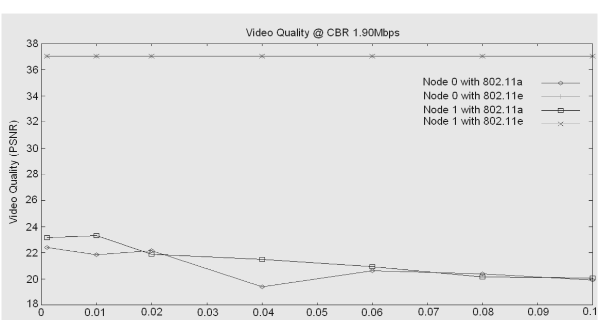

Figure 10: Error Rate Vs Video Quality

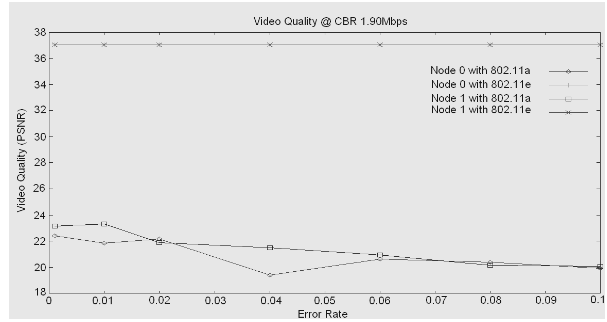

Figure 11: Error Rate Vs Video Quality

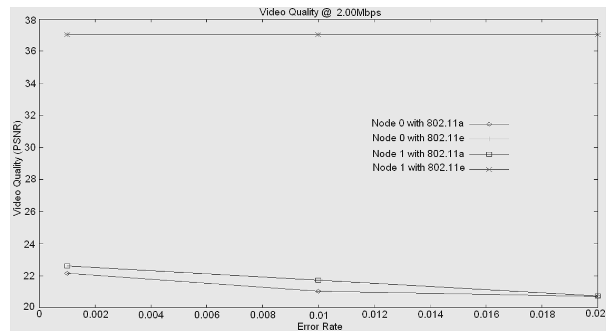

Figure 12: Error Rate Vs Video Quality

Figures 11-14 show graphs between Error rate and Video Quality at different CBR Background traffic, when two mobile nodes are receiving the same data. We have very interesting results at different CBR, with the increase in error rate in 802.11a PSNR decreases in both the nodes and the ratio of change in both nodes is almost same. There is a small fluctuation in both the nodes in Fig 10 and 11. This is because of random Uniform error pattern and random packet loss. PSNR in both nodes eventually reaches to a point where video is almost distorted but if we look at the graphs in case of 802.11e the graph shows the straight line. This is because of the same reason which we described above in first experimental setup in Figure 5 & 6 and our results confirm this fact.

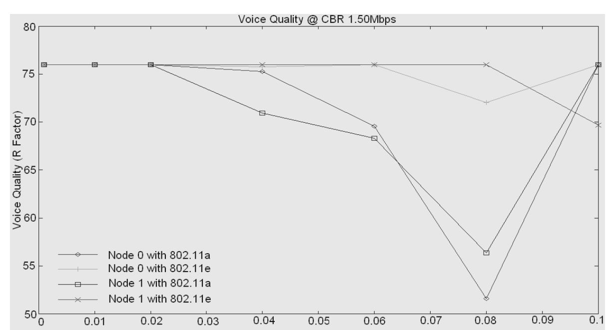

Figure 13: Error Rate Vs Voice Quality

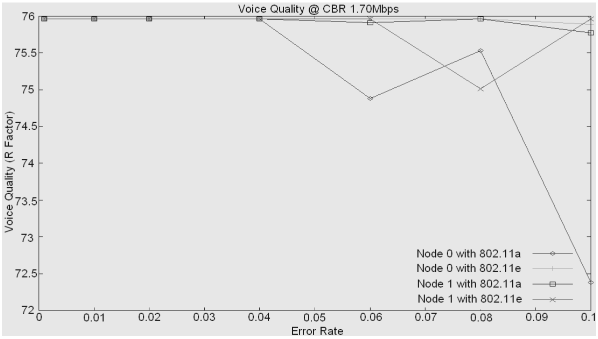

Figure 14: Error Rate Vs Voice Quality

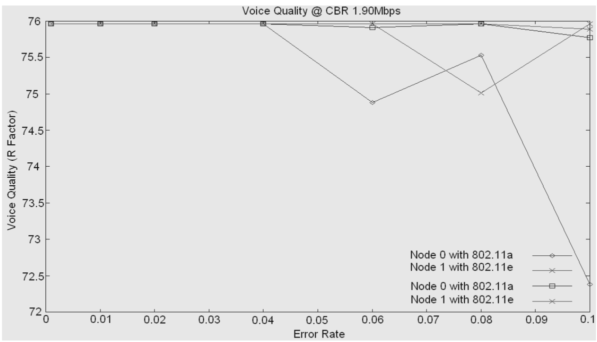

Figure 15: Error Rate Vs Voice Quality

Figures 13-15 show graphs between Error rate and Voice Quality at different CBR Background traffic. We have simulated up to 10% error rate and noticed that for low data rates quality of voice remains stable up to 7 or 8 % error rate but as soon as we increase error rate, voice quality decreases in 802.11a in both wireless nodes. In case of 802.11e there is very small change in the voice quality and generally voice quality is equal or greater than toll quality. When the CBR traffic is increased to 2.0 Mbps or above, there is a very substantial packet loss which results in a very poor voice quality.

5. CONCLUSIONS

In this paper, we studied performance evaluation of wireless 802.11a and 802.11e in an ideal environment (without errors) and also in the presence of errors in wireless network for multimedia applications. Our results show that when 802.11e QoS is used, network performance of multimedia applications is far better in infrastructure mode as compare to 802.11a. Even under heavy load conditions 802.11e performs far better than 802.11a for both voice and video traffic. This is also true for more than one mobile nodes acting as receivers. Future direction of this research could be a study of performance of multiple transmitter and receiver nodes and the performance comparison between different network topologies.

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