VoIP Implementation and Experiments on a Mobile Wireless AdHoc Network

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Abstract. We have implemented a testbed to study the performance of VoIP in wireless ad hoc networks. The ASNC (Adaptive Source Network-rate Control) scheme is used to battle packet loss by feeding MOS (Mean Opinion Score) and loss information back to the sender. Our different VoIP experiments with measurements on delay, packet loss rate and MOS have validated the feasibility and efficiency of our scheme. It further verifies the simulator we used in the initial investigation.

Keywords: Adaptive rate control, testbed implemenation, MOS, bandwidth.

I Introduction

VoIP (Voice over IP) over ad hoc networks is a promising application/technology because one can provide good quality voice communication at low cost. However, many issues need to be researched and solved; for example, improving the quality of voice. There are many reasons for the quality of voice to deteriorate. First, there are many types of interference (co-channel, adjacent channel and inter-symbol etc.) and losses (fast path loss, shadow fading and multi-path fading etc.). All these can contribute to bit errors and therefore packet drops. Many error resilient and error mitigation techniques have been investigated in recent years, and they can be classified into two basic control mechanisms. 1) ARQ (Automatic Repeat Request) [LeWi04] technique by retransmitting corrupted data frames, and 2) Forward Error Check (FEC) [QuPe05] which has the advantages of no feedback and bounded delay for real-time requirement when compared to ARQ.

Limited bandwidth in the wireless channel is another reason for the deterioration of voice quality in wireless networks. This has resulted in losing or delaying packets due to collisions in the media access layer and congestion in the network layer. Congestion control schemes and speech/channel coding are two basic technologies to resolve these problems. There are works on providing good voice quality over narrowband speech [OjLa06], on the effect of codec characteristics on the performance of wideband speech [VaDe06], and on a wideband coding architecture with applications [AhJe06]. Of particular interest is the speech coding standard AMR-WB (Adaptive

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Multi-Rate WideBand) [VaDe06] that can provide high speech quality with low coding bit rates but robust to bit loss. AMR-WB makes communication more natural, reduces listening effort, and eases speaker recognition. An adaptive rate control scheme [ZhLo06] is used to adjust the bit rate according to channel capacity.

There is also a strong interest in cross-layer design that can coordinate the functionalities in several layers. Examples are designs that combine adaptive control mechanisms with source and channel coding together, e.g., the work of [ZhTa06] uses a synthesis method to compensate wireless network weakness and then improving the quality of wireless networks. Other works include a voice MM (Multiplex-Multicast) scheme for overcoming the large overhead effect of VoIP over WLAN [WaLi05], and an E-model to study the quality of VoIP calls transmitted over the wireless medium [NaDa06]. However, there appears to have no proposal to improve voice quality in ad hoc networks to our best knowledge, until the recent ASNC (Adaptive Source-Network rate Control scheme) [ZhZh08] which has combined source rate control, network rate control and packet transmission. The performance of the algorithm, however, is validated by simulation on static networks.

There are also different testbeds developed for the study of WLAN and ad hoc networks. Hardware-fitted Modeling [FeKa00] is used to analyze the behavior of the wireless VoIP by taking hardware measurements for a small number of hardware sources, then duplicated and extended in the OPNET simulation. A large-scale testbed [LuLu02] was designed to assess several different routing protocols on a network up to 37 physical nodes. Again, there was no efficient control scheme to combat packet loss or error in ad hoc networks to our best knowledge.

This paper reports on the progress in the use of ASNC to support voice communication on wireless adhoc networks. We have since realized the ASNC scheme on a hardware-based testbed. Unlike the static networks in simulation, we have conducted experiments under different mobile scenarios. Our measurements demonstrate the effectiveness of our ASNC scheme to support VoIP on wireless adhoc networks. The results also confirmed the accuracy of our Opnet simulator.

The rest of the paper is organized as follows. Section 2 describe the testbed configuration and the network operation involving the ASNC scheme. Section 3 provides a description of our testbed implementation. Section 4 describes two mobile scenarios and discusses our measurement. Section 5 compares our measurement with simulation results. Conclusion is provided in Section 6.

2 Testbed Configuration and Network Operations

Figure 1 shows the layout of our testbed. The central ASNC scheme will be described shortly. Here, we shall first describe the functional blocks before their implementations are discussed in the next section.

a) Voice Sampling and Coding

The first block on the left performs analogue-to-digital conversion of the speech signal. The speech is sampled at 16 kHz and converted to 14 bit uniform PCM. This is implemented by a Windows Recorder with 16 kHz sampling and 16 bit representation. The uniform PCM is converted to AMR-WB code in one of the 9 rate modes (with coding bit rates from 6.6 to 23.85 kbps) for transmission over the wireless channel. DTX (Discontinuous Transmission) mode [ITU02] is also used by the VAD

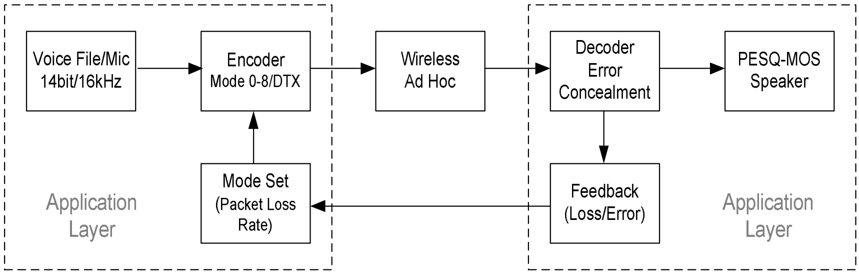


Fig. 1. Testbed Configuration

to lower bit rate requirement in the second block. More information for AMR-WB will be provided later.

b) Ad Hoc Routing

The third block is the transmission channel of a wireless ad hoc network. The MAC (Medium Access Control) protocol is CSMA/CA using DCF (Distributed Coordination Function) [WaLi05]. After the AMR-WB coding, voice data are encapsulated by the IP/UDP/RTP protocols and the ad hoc routing protocols in the network layer, and then transmitted over the ad hoc network. The intermediate nodes receive the packet and add a new IP address extracting from the packet and retransmit again in the network layer. In a wireless network, packet loss and errors can be the results of congestion, link breakage, path loss, fading, and interference. Instead of DSR (Dynamic Source Routing) [JoMa96], we use the MSR (Multi-path Source Routing) protocol [WaSh01] to implement multiple paths in order to improve network stability and the performance of network.

c) Decoding and Evaluation

The fourth block is the AMR-WB decoding. Our decoder uses CRC (Cyclic Redundancy Check) to determine packet loss according to sequence numbers, timeouts and packet errors. If a packet loss or error occurs, the receiver would use the error concealment mechanism to repair errors by filling in suitable data. After the data are decoded and played out by the sound card, we can check the perceptual quality of speech. When a transmission finishes, the PESQ procedure [ITU01] is used to evaluate the quality of speech by comparing it with the original to obtain the MOS value of the received speech.

d) Feedback and Mode Decision

When packet loss or error occurs, the sender collects this information and calculates the packet loss rate. This information is passed back to the sender using an ACK packet (e.g., at regular intervals). It then uses a mapping relationship to decide a suitable bit rate (mode) for the ANR-WB encoder. Fig. 2 is an example from our previous study [ZhZh08] in which one can determine the mode **m**n (n=0 to 8) from different combinations of MOS and packet loss rate. Apart from sending at constant rate at each mode, the encoder can also choose the DTX (Discontinuous Transmission) control to send at variable bit rates. If the lowest rate still cannot satisfy the performance, we can transmit longer packets, for example 40 ms, to reduce the overhead traffic but at the cost of increasing delay.

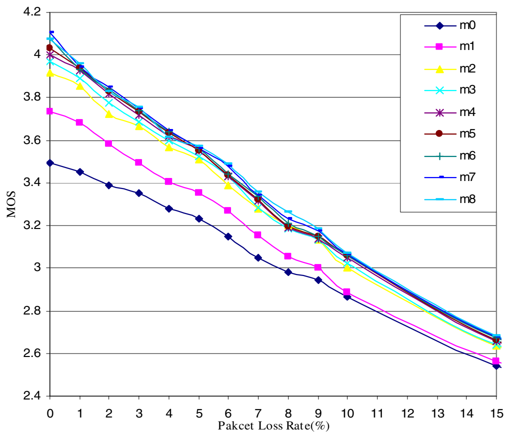


Fig. 2. AMR-WB characteristics: MOS vs. packet loss rate [ZhZh08]

The relationship among various network parameters and their effect on MOS have been studied and analyzed [ZhZh08] using the characteristics of wireless channel and AMR-WB speech coding. For example, the relationship in Fig. 2 [ZhZh08] allows us to use the rate control and packet combination to improve the transmission performance. Essentially, we use AMR-WB coding to decrease packet rate. We also utilize VAD (Voice Activity Detection) to decrease the coding rate further, and merge two or more frames together to reduce overhead of packet header, thus alleviating congestion.

2.1 The ASNC Scheme

Central to the testbed operation is the ASNC (Adaptive Source-Network rate Control) scheme introduced in [ZhZh08] which we summarized here. As evidenced from the block operations provided before, ASNC measures various parameter indicators such as packet loss rate and MOS (Mean Opinion Score). Note that the MOS is not easy to measure because it needs the original voice as reference. By making use of packet loss rate and bit rate (mode), one can get the current MOS value such as Figure 2. Vice versa, the ASNC scheme uses MOS as criteria to adjust coding mode according to the network status. When the traffic has increased to cause more packet loss, the control scheme decreases the speech coding mode. When the traffic decreases and packet loss reduces, the scheme increase the coding mode. Packet combination can be used when the payload of the speech packet is small (e.g. about 60 bytes) and header of packet is large (e.g. about 82 bytes). Thus, we make maximum use of network resources and optimize speech quality.

In summary, ASNC is a control scheme that combines source rate control, network rate control and packet combination. It controls the overall end-to-end performance by measuring the MOS value for a voice stream. We utilize feedback to improve the performance of speech transport, where status information (including packet loss rate and routing information) are provided to the source to control AMR speech coding rate and the packet transmission. Thus ASNC can exert control over the whole system in order to make maximum use of network resources.

3 Testbed Implementation

Our voice transport testbed has a flexible implementation in order to test the VoIP operation and to verify the efficacy of our ANSC scheme. The hardware and software details are provided below.

3.1 Hardware Setup

We implemented out ad hoc network on of IBM T40 and R40 laptops equipped. Each laptop is equipped with a Cisco Aironet 350 Series PCMCIA wireless card. The Cisco card is set in the ad-hoc mode, and all wireless cards use the same network name and mask so that they can be interconnected as one ad-hoc subnet without an infrastructure (i.e., without using an access point). We choose the lowest transmission power level of 1 mW in our experiment so that we can test different scenarios in as small a range as possible. The link speed is configured to 11 Mbps using Channel 1.

3.2 Software Setup

3.2.1 Testbed Operating System

We have chosen Red Hat Linux as the operation system because there is plenty of open-source software in the Linux platform. For example, the Aironet Client Utility that comes with the Cisco wireless card is a GUI (Graphic User Interface) tool used to configure the wireless card. We first use it to load the firmware of wireless adaptive card. We then use the profile manager menu to configure the transmission power, the link speed, and the network name as discussed before. We use the MSR routing protocol software [WaSh01] from Tianjin University, China because it is also based on the Linux operating system. There are also modules for measuring the quality of voice. We use the ITU P.862 software [ITU01] to get the MOS value of voice stream. We use Ethereal [Orbe04] to analyze various protocols and packet contents. Thus, we can generate, capture and measure the quality of a VoIP stream.

We also use the Linux platform to code and develop our application software for AMR-WB over IP with an adaptive control scheme. The AMR-WB coding algorithm uses the ITU-U source code. We have implemented/coded a VoIP program to generate voice streams for testing. We also use the packet generator called D-ITG [AvGu04] to generate background traffic with the VoIP stream. The commands “ITGSend” and “ITGRecv” are used to generate and to receive the UDP packets. We have also written programs to generate certain packet loss and then test the variation of MOS for different packet loss rate.

More details are provided in the next section.

3.2.2 VoIP Software

The VoIP software is the main measurement software. It consists of the sound recording/playback module, the AMR-WB coding module, the socket communication module, and the control schemes implementation module. They are described in the following sections

a) Sound Recording/Playback

In order to implement sound recording and playback, we first OPEN the sound card file and then configure the sound card using the ICNT command. As seen from the pseudo code in Fig. 3, we use either FREAD to read via a sound device to obtain recording data, or FWRITE to write sound date to drive the sound card play sound. Finally, we CLOSE the sound device. FREAD and FWRITE are blocking commands which require them to wait until the sound card finishes collecting data. They can be used to control the time of reading speech data from sound cards. Therefore they can be used for timing when we use a voice file as sending sources.

b) AMR-WB Coding

The encoder, decoder functions and PESQ are declared in Fig. 4. AMR-WB coding provided by ITU [ITU01] has two kinds of modes: ITU mode and 3GPP mode. We use the 3GGP mode because it can provide discontinuous transmission operation and bad frame indication. We compile coding and decoding codes as a static library so that other programs can call it up conveniently. In addition, we also call error concealment function when time-out for receiving occurs.

c) Socket Communication

TCP/IP protocol is used to realize communication of nodes. Since the packet data structure of the layered model is too complex to implement, socket function provided by Linux is used to encapsulate data to TCP/UDP packets and realize the node communication.

```
//open sound device
fd = open("/dev/dsp", O_RDWR);
//configure sound card parameter
arg = SIZE,CHANNELS,RATE;
ioctl(fd, SOUND_PCM_WRITE_RATE, &arg);
//recoding sound and playing
read(fd, signal, sizeof(signal));
// Playback the decoded stream
write(fd, synth, sizeof(synth));
// close the sound device*/
fclose(fd).
```

Fig. 3. Sound Recording/Playback

```
//AMR CODER signal is compressed into stream
amrcoder(FILE *f_serial, short *signal,short
      mode, short *stream,int frame)
//AMR DECODER cpprsm is decompressed into synth
amrdecoder(FILE *f_serial, FILE *f_synth,
        short cpprms, short *synth, int frame)
//PESQ measuring MOS comparing degraded file with reference file
pesqmain(char ref_file[], char deg_file[],
        long frequency);
```

Fig. 4. The Encoder, Decoder functions and the PESQ

Fig. 5. Multi-hop Scenario

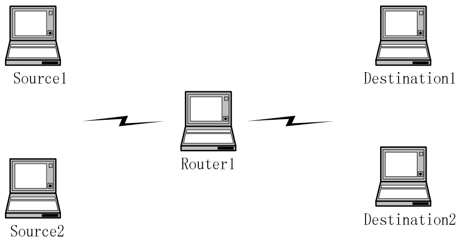


Fig. 6. Bottleneck Scenario

4 Mobile Scenarios and Their Measurements

There are two outdoor mobile scenarios from which we conducted our measurements.

(1) a multiple-hops scenario (Fig. 5): this scenario accounts for the situation that ad hoc packets need to be relayed by intermediate nodes, and will consume additional node power and network bandwidth. Unlike earlier experiments that have all nodes fixed, we move the destination node back and forth between the destination and source node positions shown in Figure 3. The node moving speed is 1.5 m/s.

(2) a bottleneck scenario (Fig.6): this scenario is used to show the effect of an intermediate node between two sources and two destinations has its packet forwarding capacity saturated. Source 1 sends a voice stream to Destination 1 and Source 2 sends background traffic to destination 2. Unlike the stationary cases studied before, our bottleneck node moves around a circle at speed 1.5 m/s.

We assembled a team of 6 students (actually 5 is enough but need additional one to help out in communication), each carrying a laptop to check the sending and receiving of signals and for recording. The laptops are held at about 1 meter above ground. Each student also carries a walkie-talkie for ease of communication over a long distance (~70 meters).

4.1 Measurements and Observations

The experiments were conducted at a data transmission rate of 11 Mbps. The client sends the speech signal using AMR-WB with 20 millisecond packet. We use the lowest transmission power of 1 mW to conduct the test with a hop distance of 70 meters. The traffic packet size is 60 bytes. In order to test the performance of speech, the voice streams and background traffic streams in both scenarios are simultaneously transmitted at the source node and receive them at the destination node. The VoIP program is used to send speech streams. The packet generator D-ITG [AvGu04] is used to generate background traffic. We use FTP on different file sizes. Other experimental settings can be found in the implementation section in Section 3. Unless specified explicitly, we use the default settings of the wireless cards.

We have also implemented the CBR (Constant Bit Rate where the coding and sending rate do not change) in order to form a benchmark for comparison with our ASNC scheme. Each measurement is repeated twice and the average of the three measurements is obtained. Some 95% confidence intervals had been measured, and were small when compared to the mean.

Below we report on different performance measures we have obtained. When comparing between the two scenarios, please note that the scales of their performance diagrams might be different.

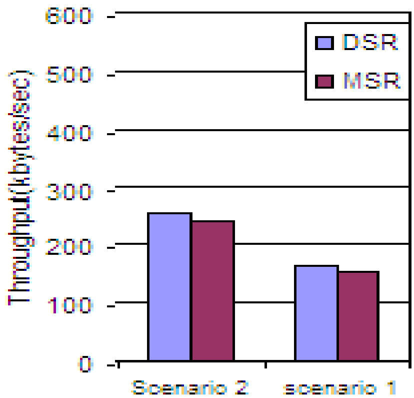


Fig. 7. Throughput Performance in Mobile Environment

4.1.1 Throughput

Figure 7 presents the throughput performance of the mobile ad hoc network using DSR and MSR. When comparing the 2-hop networks (Scenario 2) and 3-hop networks (Scenario 1), one sees that the throughput of the network decreases when the number of hops increases. Because each node shares the same wireless broadcast channel and relay nodes also consumes bandwidth to forward packets; the more hops, the less the available bandwidth is. Although about the same, the DSR has a slightly better performance than MSR because MSR is a more complex algorithm.

4.1.2 Packet Loss Rate

Figure 8 presents the packet loss rate performance using the MSR operation. CBR is compared to ASNC in each scenario. As seen in Scenario 1 (Figure 8a), the packet loss rate under CBR (CBR(mout)) is increasing exponentially with regard to the packet arrival rate. The performance of ASNC is similar but much reduced. For example, the loss rate is 16% at 400 packets/sec as opposed to 24% for CBR. This demonstrates the capability of ASNC in combating/adapting to the channel environments including the bandwidth fluctuation due to channel loading. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 8b), but the loss rate is decreasing. For example, at 500 packets/sec, ASNC loss rate is reduced

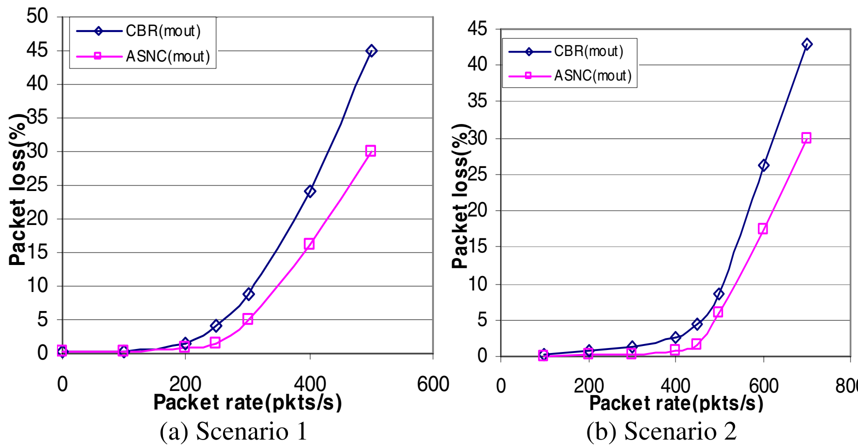


Fig. 8. Packet Loss Performance

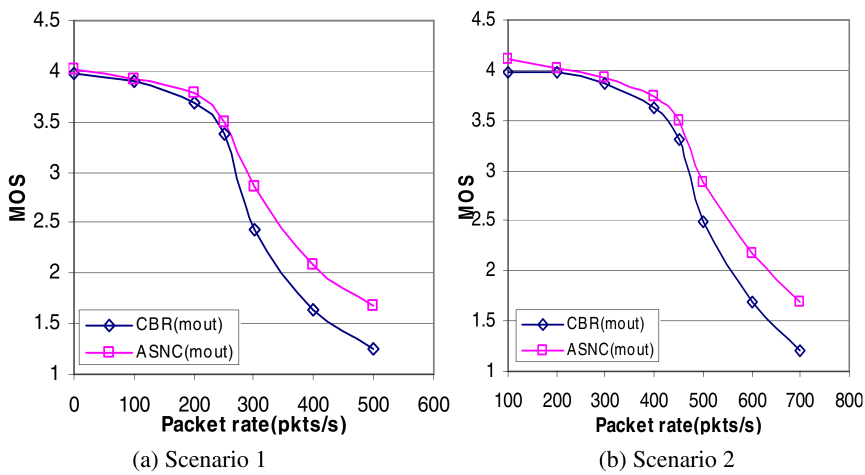
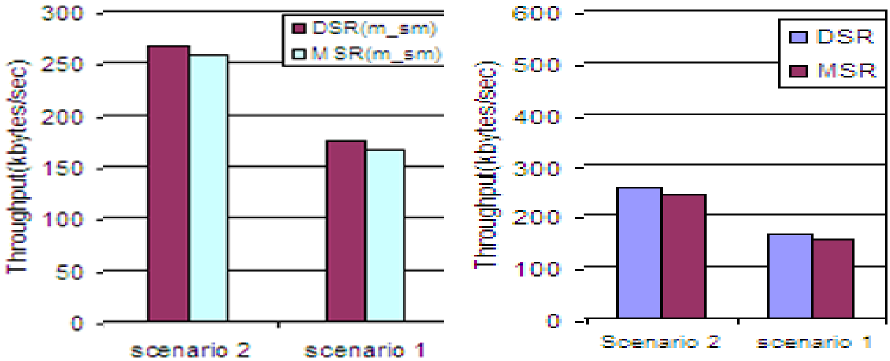


Fig. 9. Speech Quality Comparison

from 30% in Scenario 1 to 6% in Scenario 2. This is because Scenario 1 has more hops than Scenario 2.

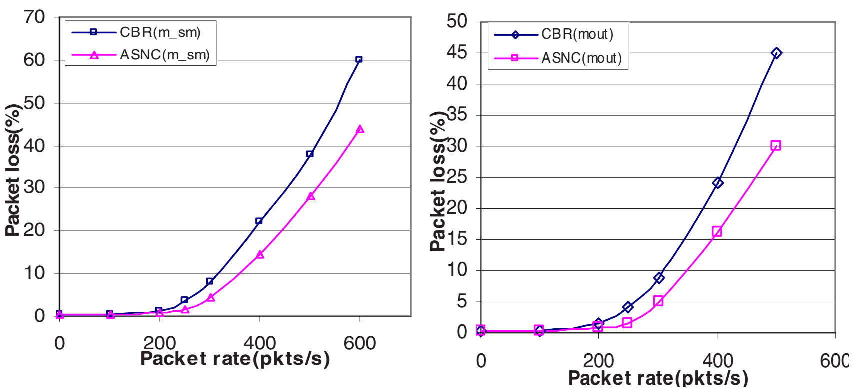
4.1.3 Speech Quality

Figure 9 presents the speech quality performance using the MSR operation. CBR is compared to ASNC in each scenario. As seen in Scenario 1 (Figure 9a), the MOS under CBR (CBR(min)) is decreasing with respect to the traffic packet rate. The performance of ASNC is similar but much increased because ASNC is capable to reduce traffic. For example, the MOS is 2.9 at 300 packets/sec is opposed to 2.4 for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 9b), but the MOS is better. For example, at 500 packets/sec, ASNC MOS is reduced from 1.7 in Scenario 1 to 2.9 in Scenario 2. By inspection, Scenario 1 has the smallest MOS because it has more hops (3 hops) than Scenario 2.



(a) Simulation (b) measurement

Fig. 10. Throughput Performance Comparison



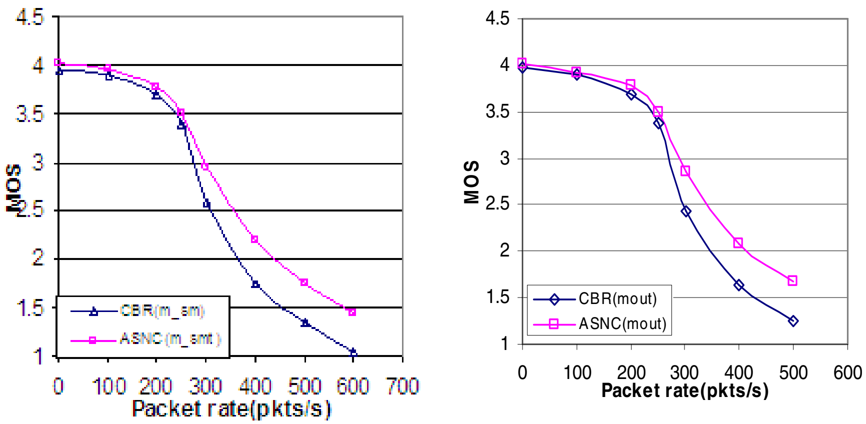
(a) Simulation (b) Measurement

Fig. 11. Loss rate performance comparison under Scenario 1

5 Verification of Simulations

We have actually conducted more simulations on mobile networks since our report on static networks last time [ZhZh08]. So it would be interesting to see how they compare.

Note that we only present some of the comparisons here because their comparisons are quite close. For examples, Fig. 11 shows that the loss rate performance is 14% at 400 packets/sec for simulation and is close to 16% for measurements. Fig. 12 shows that the MOS at 500 packets/sec is 1.4 for simulation while it is 1.3 for measurement. Similar observation can be found in the throughput of Fig. 10 (note the scales are different). Some other comparisons show even closer matching and therefore are not presented. So indeed our simulation model previously can capture the real network scenarios closely. We can use them in the simulation of a bigger network.



(a) Simulation (b) Measurement

Fig. 12. MOS Performance under Scenario 1

6 Conclusion

In this paper, we have reported on the implementation and new measurements results of our mobile VoIP adhoc networks. Our measurements demonstrated that our ASNC scheme can improve the quality of voice transport over a mobile wireless ad hoc network. We have also validated our simulation results which allow us to use them with more confidence for larger networks in future.

Acknowledgment

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