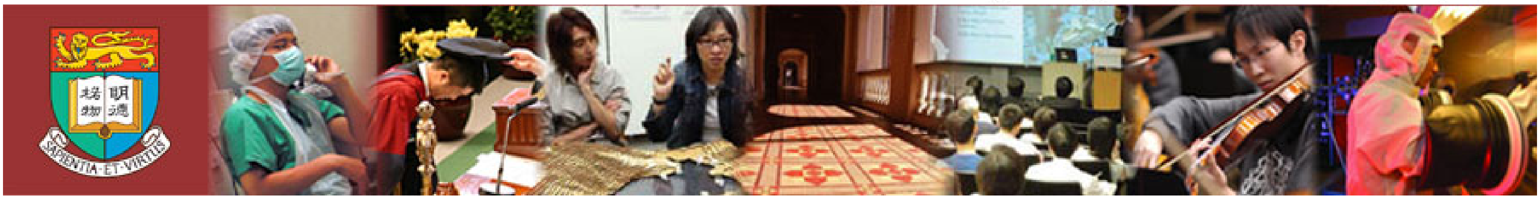

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# Transporting Audio over Wireless Ad Hoc Networks: Experiments & New Insights

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*Abstract- Current efforts on ad hoc wireless network research are focused more on routing and multicasting protocols. However, there is an increasing need to understand what sort of media could be transported over wireless ad hoc networks other than data. Existing research on multimedia wireless communications often addresses broadband wireless networks with a connection-oriented backbone. In this paper, we address the possibility of transporting audio traffic over wireless ad hoc networks. We examine the impact of wireless multi-hop links on audio data relay and how the audio quality at the receiver is affected. In particular, we examine communication parameters such as latency, jitter, packet loss, and their impact on perceived audio quality.*

## I. MOTIVATION

Multimedia technology is commonly found in most personal computers and laptops today. It has allowed computers to display video and playback stored audio. With the interconnection of computers to the Internet, one is now able to transmit and receive audio and video information from remote hosts. Many audio and video conferencing tools have also evolved, allowing multiparty communications and collaborations to be possible.

Several standardization efforts have evolved to support multimedia communications. The ITU-T has introduced a set of "H series" standards that cover signaling protocols, media streaming, security, and encryption. ITU has also established a set of media encoding standards to support voice, such as G.711 PCM, G.726 (ADPCM), and H.728 (LD-CELP)[2]. The IETF, on the other hand, has also made several efforts in the areas of session protocols (RFC 2543 and RFC 2327) and media transport (such as RTP[1], RTCP[2], and RTSP[3]). However, most existing standardization and research efforts[7][8][9][11] on multimedia communications and transport are devoted to wired networks, not wireless. In particular, issues related to transport of multimedia traffic over ad hoc wireless networks have not been investigated in greater depth. Hence, it is the purpose of this paper to examine the issues associated with sending audio over multi-hop wireless links.

This paper is organized as follows. Section 2 provides a brief background on audio coding technologies and routing for ad hoc wireless networks. This is followed by a narration of our experimental hardware and software setup in Section 3. Section 4 provides an evaluation of the experimental results obtained. Finally, section 5 discusses future work and concludes this paper.

## II. BACKGROUND

### 2.1 Audio Encoding Technology

Audio technology has advanced tremendously over the last few years. Digital audio coding, commonly known as digital audio compression, allows the minimization of storage space and channel bandwidth requirements for audio data. MPEG Layer 3 or MPEG-2 AAC (Advanced Audio Coding) exploits the limitations of the human ear to achieve a size reduction by a factor of 12 with little or no perceptible loss of audio quality. The encoding process transforms the digital audio data (in a WAVE file) into a highly compressed form called bit stream (i.e., coded audio data). To extract from bit-stream format back to WAVE format, decoding is needed.

A table outlining the unique features of nine audio formats is shown in Table 1. There are pros and cons related to different formats that use different encoding methods. Some require very low bit rate, such as ".ra", and ".vox" formats. Generally, the higher the compression factor, the lower is the bit rate. Audio quality is affected by the coding method used. For the purposes of our experiments, the ".wav 16-bit PCM" audio format was chosen because it has very high audio quality. Using a high quality audio format ensures that we

[1] RFC 1889 – RTP: Realtime Transport Protocol
[2] RFC 1889 – RTCP: Realtime Transport Control Protocol
[3] RFC 2326 – RTSP: Realtime Streaming Protocol

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can gauge our output against a perfect sample. This is evident since it has a compression factor of 1:1, which means no compression. Thus, in our experiments, we were assured that the purest and highest quality audio was being transmitted. One disadvantage of transmitting audio using the ".wav" format is its large file size which is a result of a low compression ratio. Another shortcoming is that it does not support streaming. Hence, it takes a longer time to transmit a message using ".wav" file than other formats.

| Audio Format | 16-bit PCM | G.711 Mu-law | 32Kbps MPEG-1 | IMA/DVI ADPCM | GSM | Real Audio v1.0 |
|---|---|---|---|---|---|---|
| File extension | .wav or .aiff | .au | .mpa or .mp2 | .wav | .gsm | .ra |
| Rate (Kbps) | 128 | 64 | 32 | 32 | 13.2 | 8 |
| Size /min | 960 | 480 | 240 | 240 | 96 | 59 |
| Compression Factor | 1:1 | 2:1 | 4:1 | 4:1 | 10:1 | 16:1 |
| Sound Quality | 5 | 4 | 4 | 3 | 2 | 1 |
| Unix player | Yes | Yes | Yes | Some | Yes | Yes |
| Windows player | Yes | Yes | Yes | Yes | Yes | Yes |

Table 1: 8KHz Mono Audio Formats.

### 2.2 Advances in Ad Hoc Mobile Routing

Ad hoc wireless networks are different from cellular-based networks since there are no fixed radio base stations. Routes in an ad hoc wireless network consist of multi-hop wireless links. In addition, each node in an ad hoc wireless network is mobile and acting as a router, relaying packets sent by others towards their intended destinations.

Several routing protocols have been proposed to IETF. Compared to the routing protocols used in today's Internet, ad hoc routing protocols need to deal with mobility, route convergence, constrains in power, limited bandwidth and varying link quality. A review and comparison of existing ad hoc routing protocols had been reported in [4]. Basically, there are three categories of ad hoc routing, namely: (a) on-demand, (b) table-driven, and (c) hybrid. On-demand protocols attempt to discover a route on-the-fly based on needs by the source. Table-driven protocols, however, rely on periodic propagation of route updates to maintain a consistent and up-to-date routing table. Mobility is reflected by link changes and hence results in route updates. The hybrid approach combines features of on-demand and table-driven routing to address scalability issues.

Recently, some researchers have successfully demonstrated the correctness and practicality of their protocols. An ad hoc wireless networking testbed[5] based on associativity-based routing (ABR)[3] had been implemented. ABR differs from other protocols in that it advocates for stable and long-lived routes instead of shortest path as the most important routing metric. The argument is that a shorter-hop route can easily be broken in an ad hoc mobile network since each node is mobile.

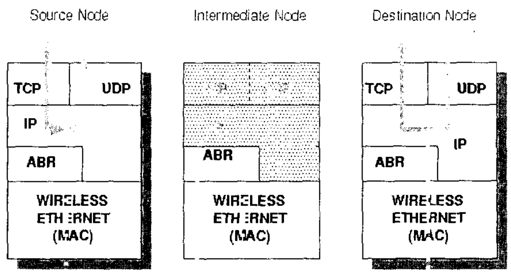


**Figure 1: The Concept of Multihop Packet Forwarding in ABR Routing.**

## III. EXPERIMENTAL ENVIRONMENT

### 3.1 Systems Hardware

Having discussed audio technologies and ad hoc wireless networks and routing, we present our experimental hardware and software setup. Our ad hoc wireless network testbed consists of several laptops (Dell, IBM, and Compaq) equipped with Lucent WaveLAN PCMCIA cards. The radios have a data rate of 2 Mbps and operate at 2.4 GHz. The radio coverage ranges from 50m (indoor) to 200m (outdoor), depending on the environment. The channel access protocol used is the CSMA/CD protocol. The laptops in used are capable of handling audio since the necessary audio hardware is built into these laptops.

In ABR, nodes learn about the connection stability of links over time and space, with the use of beacons. These periodic beacons not only identify the nodes that send the beacons but also information about the power life of these nodes and their degrees of association stability with one another. If a node constantly records beacons received from a specific mobile node, then even if that node is moving, it still retains good connectivity relationship, i.e., it can be classified as a good neighbor. This, therefore, summarizes the fundamental concept behind ABR routing.

### 3.2 Communication & Audio Software

In addition to hardware, all the laptops used in our experiments were running Linux OS with a modified systems kernel. The ABR ad hoc routing software is incorporated into this kernel, as shown in Figure 1. When the laptop boots, it also executes the ABR protocol in the background. Each laptop is capable of communicating directly to a neighboring laptop or to a remote laptop via other laptops. No radio base stations are used in our testbed. Linux allows us access to open source code and also provides us with a multi-tasking windowing environment.

Performing debugging work in Linux is also relatively easy. The "Hawaii-Five-O" wav audio file that we used for our experiments is available from the public domain. Each laptop has a copy of this file. The appropriate audio device drivers are installed in each of these laptops so that they are capable of playing audio files from disk or from the network. We wrote additional client/server code to send and receive audio data over ad hoc wireless links. Our code subsequently writes received audio data to /dev/dsp to produce sound.

### 3.3 Ad Hoc Audio Testbed Topology

We are interested in investigating the quality of audio transmission over an ad hoc wireless route consisting of different route length. Hence, we have different route configurations setup prior to commencement of audio transmission. Figures 2 and 3 show the 2-hop route setup. The distinction here lies in the locality of the intermediate mobile node. We have also established a 1-hop "near" and "far" scenarios. The former refers to two ad hoc mobile nodes separated by a distance of 1.5 meters. The latter, however, refers to a spatial node separation of 30 meters. Due to constraints in manpower and resources, we are unable to perform experiments beyond 2 hops. However, we did include node mobility into our experiments, which will be discussed later.

### 3.4 Parameters of Interest

The quality of audio transmitted over multiple wireless links is subject to a variety of factors. In particular, *latency*, *jitter*, and *packet losses* are important parameters. In traditional Public Switched Telephone Networks, the round-trip latency for domestic calls is virtually always under 150 milliseconds. At this level, the latency is not noticeable to most people. Many international calls (especially calls via satellite) will have round-trip latency figures that exceed 1 second, which can be very annoying to users. Hence, we need to know the amount of latency incurred over ad hoc wireless networks. Jitter refers to the variations in latency. As variability in the packet arrival rate increases, the audio will begin to sound garbled. Typically, network components compensate for jitter by using buffers. Jitter buffers store incoming packets and send them out in a constant stream. It would be important to know how much jitter is introduced as audio data is transported over ad hoc wireless networks.

Excessive packet losses can also affect the quality of received audio. Typically, more than 5% lost packets will annoy users based on the ability of human perception. This amounts to one out of twenty consecutive packets. Hence, we need to know the severity of packet losses in ad hoc wireless routes. In summary, we shall discuss our experiments in relation to the above-mentioned parameters.

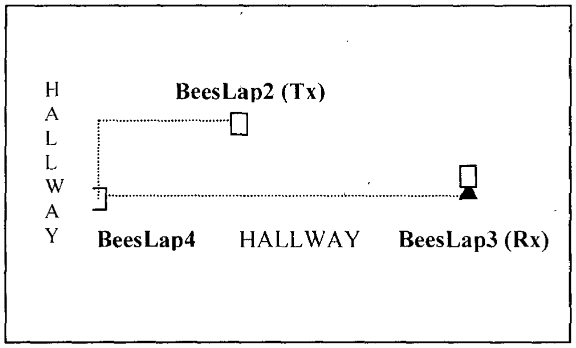


**Figure 2 Our 2-hop Audio over Ad Hoc Wireless Network Testbed Setup – Configuration 1.**

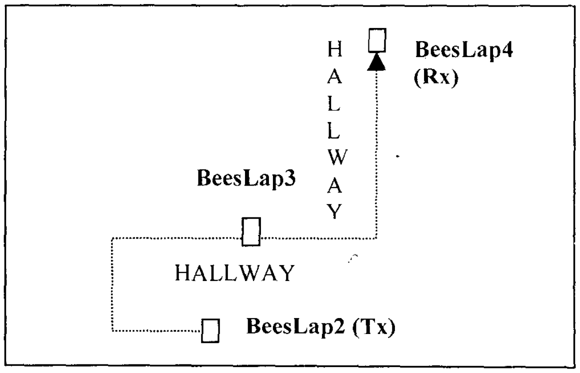


**Figure 3 Our 2-hop Audio over Ad Hoc Wireless Network Testbed Setup – Configuration 2.**

# IV. RESULTS & OBSERVATION

## 4.1 1-Hop "Near" Audio Experiments

We setup the 1-hop "near" configuration with laptops beaconing at one second intervals. All trials for 1-hop audio transmission in near proximity resulted in less than 5% packet loss w.r.t. the total packets transmitted. Statistically, very few packets were lost. In fact, only 1.48% of the packets were lost. Detailed results are shown in Table 2. Thus, from a human perceived quality standpoint, the audio transmission was of good quality.

| Attempts | Packet Loss Rate (%) |
|---|---|
| 1 | 1.56 |
| 2 | 1.04 |
| 3 | 1.56 |
| 4 | 1.55 |
| 5 | 1.55 |
| 6 | 1.55 |
| 7 | 0.69 |
| 8 | 2.77 |
| 9 | 3.67 |
| 10 | 1.81 |

**Table 2 Percentage of Packet Loss for 1-hop Audio Transmission.**

Figure 4 shows the jitter of packet arrivals. Most packets arrive with zero jitter, except for a few samples. However, they are too insignificant to be detected by the human ear.

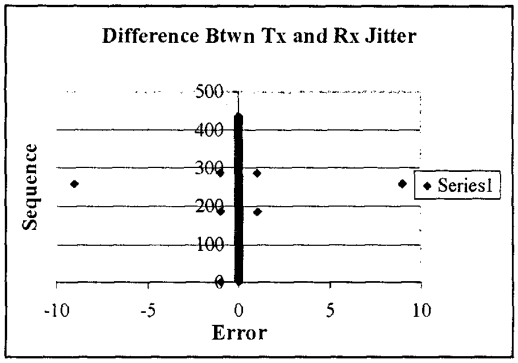


**Figure 4 Jitter as a function of audio packet sequence number for two ad hoc mobile nodes in close range.**

## 4.2 1-Hop "Far" Audio Experiments

Similar to the above scenario, all trials for 1-hop transmission in far proximity resulted in no packet loss over 5%, as shown in Table 3. The average packet loss is 1.57%. This percentage is very close to the short-range "near" experiment. Thus, it is concluded that quantitatively, the audio quality for transmission with a "far" separation of transmitting and receiving units is good.

| Attempts | Packet Loss rate (%) |
|---|---|
| 1 | 2.58 |
| 2 | 1.14 |
| 3 | 1.37 |
| 4 | 0.69 |
| 5 | 1.14 |
| 6 | 1.37 |
| 7 | 1.59 |
| 8 | 1.14 |
| 9 | 1.81 |
| 10 | 1.81 |

**Table 3 Percentage of Packet Loss for 1-hop Far Audio Transmission.**

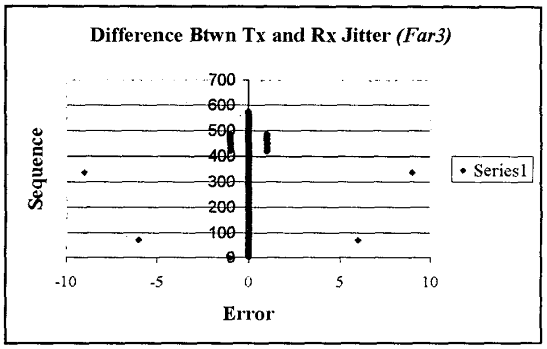


**Figure 5 Jitter as a function of sequence number for "far" 1-hop audio transmission.**

Figure 5 shows the jitter results. Compared to the "near" scenario, the presence of jitter has increased. Out of 650 samples sent, 72 samples experienced jitter during audio transport. Nonetheless, the perceived audio quality remains good.

## 4.3 2-Hop Audio Experiments

All trials for 2-hop audio transmission resulted in considerable packet loss. As shown in Table 4, 80% of the trials resulted in packet loss well over 30%. The average packet loss for the two-hop experiment was 31.87%. Thus, it was concluded that the audio quality was very poor.

| Attempts | Packet Loss Rate (%) |
|---|---|
| 1 | 26.33 |
| 2 | 52.92 |
| 3 | 41.61 |
| 4 | 16.06 |
| 5 | 32.46 |

**Table 4 Percentage of Packet Loss for 2-hop Audio Transmission.**

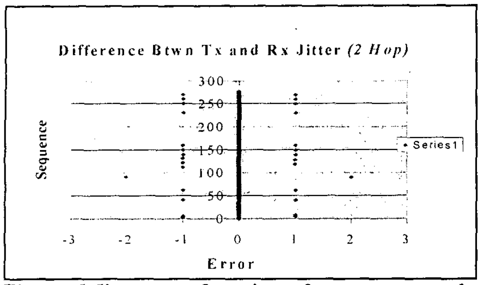


**Figure 6 Jitter as a function of sequence number for 2-hop audio transmission.**

Figure 6 shows the jitter for 2-hop audio transmission. Note that jitter is present and audio packets experienced up to a maximum of 2ms in jitter. Also, 35% of the received packets experienced jitter of 1 ms.

### 4.4 Audio Experiments with Mobility[4]

All trials for 1-hop transmission with mobility resulted in no packet loss over 5% of the total packets transmitted. The average packet loss for the one-hop mobile experiment was 4.38%. The effect of mobility on audio quality is reflected by an increased in the average packet loss. This is almost 3 times the amount experienced in close range transmission. However, the audio quality was still good.

| Attempts | Packet Loss Rate (%) |
|---|---|
| 1 | 4.33 |
| 2 | 4.33 |
| 3 | 4.85 |
| 4 | 3.29 |
| 5 | 2.94 |
| 6 | 3.79 |
| 7 | 3.12 |
| 8 | 3.98 |
| 9 | 4.16 |
| 10 | 4.68 |

**Table 5 Percentage of Packet Losses for 1-hop Audio Transmission.**

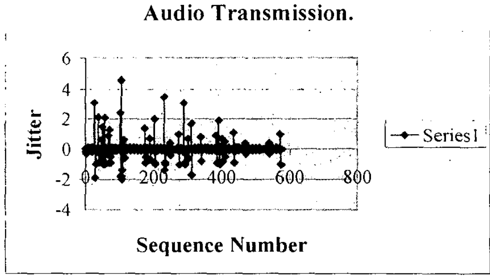


**Figure 7 Jitter as a function of sequence number for 1-hop mobility scenarios.**

4 By mobility we mean with one node moving towards and subsequently away from the other node.

Jitter performance for 1 hop with mobility is shown in Figure 7. With mobility, jitter values vary considerably. Variations in propagation distance and locality affect packet jitter. Hence, to deliver quality audio, compensation for jitter is necessary.

## V. DISCUSSION

### 5.1 Audio Communications Software

We developed the software needed for our experiments. The first version of our audio code basically loops back the transmitted audio and plays it back at the transmitter. We did this to confirm which audio we are transmitting and also the quality of the original audio. This, however, causes huge delays when it comes to measuring audio transmission time. In addition, garbled sound quality is sometimes observed. To eliminate gaps in the audio transmission, our audio code was altered to stop the audio playback at the transmitter. The receiver code was, however, left intact. Remarkably, once this change was made, the transmission worked well and simulated a real-time audio transmission scenario.

In the Linux environment, only one process can access the DSP device at any one time. Attempts to do so by other processes will be blocked. We also observed that feeding the audio data to /dev/dsp too slowly will result in pauses in the sound output.

### 5.2 Audio Experiments

To run the audio subjective test, the transmitting computer must disable audio playback to avoid causing a delay due to buffering conflicts in the sound card. However, the receiving computer must be able to play the audio back so that the sound can be evaluated by listening to it. During our qualitative tests, both the transmitting and receiving computers have their sound play functions turned off.

| Experimental Run | Packet Loss Rate (%) | Quantitative QoS |
|---|---|---|
| 1 Hop Close | 1.48 | Excellent |
| 1 Hop Far | 1.57 | Excellent |
| 2 Hop | 31.87 | Poor |
| 1 Hop with Mobility | 4.38 | Good |

**Table 6 Overall Comparison of Experimental Results.**

Qualitatively as shown in Table 6, the sound in the 1-hop "near" transmission scenario was excellent, i.e., almost flawless. This is followed by the 1-hop "far" transmission, 1-hop with mobility, and the 2-hop transmission. The sound became garbled for the 2-hop transmission. Generally, our human ear is not sensitive enough to distinguish subtle differences in sound quality and this made it difficult for us to accurately distinguish between the different experiments run.

### 5.3 Impact of Beaconing on Sound Quality

Recall that ABR protocol uses beacons is changed to reflect a decrease in beaconing rate i.e., 1s → 5s → 15s, the sound quality is dramatically improved. It actually changes from unintelligible garble to intelligible audio. In fact, the perceived sound quality was better for the 2-hop 15-second beaconing case than that of a 1-hop case with 1 second beaconing interval. This may imply that for ad hoc mobile multimedia systems, beaconing intervals should be wisely chosen unless beacons are sent out over a separate control channel. Currently, we have seen the introduction of multi-band radios in mobile systems. Hence, beaconing could well be sent over control channels.

### 5.4 Impact of Device Heterogeneity on Audio Quality

The impact of sound quality is directly affected by the capabilities of mobile devices used to send and receive audio data. For this particular experiment, the testbed consists of powerful laptops capable of supporting high sound quality. This was verified by performing a close-range one-hop transmission. It was noted that the audio quality sent and received correlate almost exactly. The 2Mbps wireless data rate is more than sufficient to support the audio codec bit stream requirement of 128Kbps. In fact, our earlier experiments on communication performance of ABR[5] reveal that at two- and three-hops, the throughput was at an average of 600Kbps and 350kbps respectively.

A finding we felt worth reporting is the incompatibility of computer devices and its impact on audio quality. For the scenario where the source is a high-end IBM laptop and the receiver an inferior COMPAQ laptop, we discovered that the DSP in the COMPAQ is relatively poor in handling and playing back received audio (due to lower memory and MIPS). Further research is, therefore, necessary to support audio over heterogeneous ad hoc mobile platforms.

## VI. CONCLUSION

In this paper, we present the insights of our experiments associated with transmitting and receiving audio information over ad hoc wireless networks. This is a relatively new research area. Generally, as delay, jitter or the number of lost packets increases, the audio quality decreases. Changes in audio quality are more realizable when it comes to packet loss and large packet jitter. Beaconing intervals and device heterogeneity will influence received audio quality. Future work includes exploration of new ad hoc voice applications, testing with using audio sources encoded in different formats, and mobile devices equipped with multiband radios.

| | |
|---|---|
| G.723.1 (1995) | Algebraic Code Excited Linear Prediction. Operates at 5.3 and 6.4 Kbps |
| G.726 (1990) | Adaptive Differential Pulse Code Modulation (ADPCM). Operates at 16, 24, 32, and 40 Kbps |
| G.728 (1994) | Low Delay Code Excited Linear Prediction (LD-CELP). Operates at 16Kbps |
| G.729 (1995) | Conjugate Structure Algebraic Code Excited Linear Prediction (CS-ACELP) Operates at 8Kbps |
| G.729A (1996) | Uses CS-ACELP Operates at 8Kbps. |

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