

Performance of a Bluetooth IP Network for Streaming High Quality Audio

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Abstract

An 'Internet Protocol' (IP) network established over Bluetooth affords higher practical throughput compared to the 'Synchronous Connection Oriented' (SCO) physical link allowing the usage of high quality audio codecs such as the 'Motion Picture Expert Group-1 Layer 3' (MP3) and 'Ogg Vorbis' audio codecs. The Bluetooth IP network by default sets the 'Logical Link Control and Adaptation Protocol' (L2CAP) layer to infinite retransmissions effecting a trade-off between audio quality and audio playback continuity. This paper studies the attributes of the Bluetooth IP network for streaming stored high quality audio.

1. Introduction

Due to the increasingly widespread usage of IP enabled devices, the Bluetooth 'Personal Area Networking' (PAN) profile allows the establishment of the TCP/IP stack atop an existing Bluetooth stack to facilitate IP communications[2]. Lossy compressed audio codecs such as MP3 and 'Ogg Vorbis' necessitate smaller bandwidth utilizations compared to the standard wave (wav) codec[4]. Therefore, lossy compressed audio files enables high quality audio streaming within the confines of the Bluetooth link which theoretically amounts to 723.2 Kbps[1].

2. Simulation Setup

The setup comprised two 'Red Hat Linux 9' laptops equipped with Bluetooth class 1 adapters. BlueZ was utilized as the Bluetooth stack while 'Java Sound' performed audio decoding. A stationary audio server streamed UDP audio packets to a mobile audio client under different simulation environments namely:

Ideal: Without interference.

Frequency Spectrum Interference: A stationary IEEE 802.11g access point was situated at a distance of 1 meter from the audio server constantly performing a file transfer (30 Kbyte/s) to another IEEE802.11g enabled laptop located at a distance of 4 meters from the audio server towards the direction of the audio client.

Movement Interference: The audio client was in random directional motion at a rate of 1 meter every 2 seconds within a radius of 12m from the audio server.

The L2CAP layer provides infinite retransmissions of packets received in error within a Bluetooth IP network through the 'Bluetooth Network Encapsulation Protocol'(BNEP)[3].

3. Performance Metrics

Throughput: The data in figure 1 assessed audio data within the UDP packet payload. Measurements were averaged across a distance that ranged from 1m to 12m. The probability of the 5000 byte packets being corruptible while being streamed through air were lower due to shorter air stream time durations. This lead to less retransmissions at the Bluetooth baseband layer and higher application layer throughput. However, the 56500 byte packets maintained a higher average throughput even though the probability for corruption was higher. The extra bandwidth dedicated to transmitting the additional headers of the 5000 byte packets were a greater influencing factor in reducing throughput compared to air stream time. The highly efficient segmentation and reassembly process at the L2CAP layer enabled the 56500 byte packets to maintain high average throughputs indicating that this process does not greatly influence throughput reduction.

Inter Arrival Duration: This metric calculated the difference in the first bit's arrival time between packets. Referring to figure 2, the DH5, DM5, DH3 and DM3 ACL modes yielded smoother audio playback compared to the DH1 and DM1 ACL modes due to the shorter amount

ACL Mode	Average throughput (Kbps)		Percentage Reduction of Average Throughput			
	Ideal Environment		Movement Interference Environment		Frequency Spectrum Interference Environment	
	56500 byte packet	5000 byte packet	56500 byte packet	5000 byte packet	56500 byte packet	5000 byte packet
DH5	619.3	596.1	4.66%	9.14%	9.98%	10.86%
DM5	447.4	434.4	2.68%	5.44%	3.27%	5.03%
DH3	519.6	501.4	4.17%	6.29%	3.58%	5.46%
DM3	357.0	349.9	2.47%	3.15%	3.04%	4.97%
DH1	165.4	160.0	2.94%	3.75%	2.82%	4.64%
DM1	105.5	102.9	2.47%	3.32%	1.78%	4.07%

Figure 1. Throughput comparison

ACL Mode	Average Inter Arrival Duration (msec)		Percentage Increase for Average Inter Arrival Duration			
	Ideal Environment		Movement Interference Environment		Frequency Spectrum Interference Environment	
	56500 byte packet	5000 byte packet	56500 byte packet	5000 byte packet	56500 byte packet	5000 byte packet
DH5	721	66	3.19%	3.03%	4.85%	4.12%
DM5	1023	93	1.47%	1.08%	2.15%	1.56%
DH3	874	80	2.97%	1.25%	3.20%	2.54%
DM3	1266	113	1.82%	0.88%	2.84%	1.14%
DH1	2784	265	2.23%	1.16%	0.47%	0.45%
DM1	4344	390	1.63%	0.51%	0.83%	0.92%

Figure 2. Inter arrival duration comparison

of time for the baseband layer to complete retransmissions. The inter arrival durations were consistent presenting a non-bursty stream. Within the interference environment, inter arrival duration only slightly increased negligibly affecting audio playback smoothness. 5000 byte packets featured higher durability to interference due to the shorter retransmission times for packets in error.

Initial Latency: Figure 3 refers to the average initial latency across all environments at a distance of 1m for 4MB audio files. The difference between the first packet's arrival time and the time whereby the receiver's audio player could maintain continuous audio playback was measured. The distinctive longer initial latencies of the DM1 and DH1 ACL modes proved to be unsuitable for audio streaming applications.

Initial Buffer Length: The buffers were measured as multiples of packet payload lengths and considered the lengths required to maintain continuous audio playback. Across all environments and distances, the initial buffer lengths for the ACL modes omitting the DM1 and DH1 ACL modes were 56500 bytes and 60000 bytes for the 56500 byte packets and 5000 byte packets.

Perceptual Audio Quality: End users rated the resemblance of the steamed audio file to the original with a 10 rating indicating an ideal match. Referring to figure 4, audio data was unable to be decoded when bit error rates were larger than 0.001 due to the codec framing and se-

ACL Mode	Average Initial Latency (msec)	
	56500 byte packet	5000 byte packet
DH5	868	919
DM5	1283	1410
DH3	1059	1197
DM3	1512	1603
DH1	33879	45354
DM1	134065	222526

Figure 3. Initial latency comparison

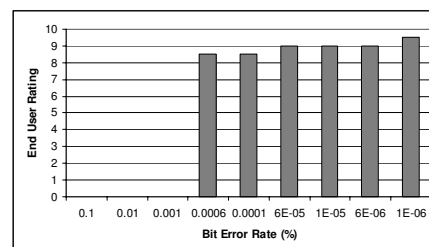


Figure 4. Perceptual audio quality

quencing information being disrupted. For bit error rates smaller than 0.01, the decoder was able to recover audio playback but at distorted levels.

4. Conclusion

The Bluetooth IP network is a feasible medium to stream stored high quality audio. The optimum setting includes the 56500 byte packet with a similar sized initial buffer length. The reliability of the L2CAP layer while at first glance may seem to impede audio continuity by having to infinitely retransmit packets in error actually improved audio quality. The retransmissions at the baseband layer were efficient enough to provide smooth audio continuity as seen by the short inter arrival durations. Bluetooth is durable enough to withstand interference environments with only minor performance degradations. The degradations under interference is not visible to the end user as long as the optimum settings are chosen.

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