Audio Transmitting on Mobile Phone by Using UDP-Lite Protocol

Eun-Seok Ryu
Department of Computer Science and Engineering, Korea University
esryu@os.korea.ac.kr

Chuck Yoo
Department of Computer Science and Engineering, Korea University
hxy@os.korea.ac.kr

ABSTRACT

As wireless data network's usage increase highly, many studies introduce multimedia data streaming techniques on mobile phone in wireless condition. To study these techniques, firstly we need to understand characteristics of wireless network by data transfer experiment. So, we set up and examined multimedia streaming condition on wireless network. And then, we investigate more effective error recovery method to transmit EVRC audio with video based upon understanding of wireless network characteristics. In addition to this research, this paper suggests use of UDP-Lite protocol to deal with errors on wireless network. After all, this article shows the way improving quality of EVRC sound on mobile phone.

1. INTRODUCTION

In general, error probability in wire network has been known statistically below 1%, but in actual test on wireless network, error probability was 5%～10% (Based on Korean environment). By this relatively high error rate, multimedia data streaming needs to error resilient codec.

In system designing, buffering time has to be reduced to support real time feature, then server can not have enough time to retransmit erroneous data. So, studies have focused on error recovery methods. To put it more concretely, Error correction/recover code was inserted in prepared data in server part, and packet was transmitted to client. And client tries to minimize data loss using those error correction codes. But, environment of these works was mainly powerful PC client platform, so it has some defects (e.g. complicated processing and redundancies). Moreover, error resilient codec which is not specified codecs endeavors to recover insensitive data. At last, several studies have been made on transport layer, and UDP-Lite protocol was designed.

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It's a best-effort protocol that adds multiplexing and optional checksum to IP. Its flexible checksumming scheme allows corrupted data to be transmitted to the application. From these studies, it was known that the quality of the radio links is not as important when using UDP Lite. To put it plainly, an application which can cope with damaged data would benefit from the UDP Lite protocol instead of UDP in a wireless network environment.

In this paper, we designed and implemented multimedia streaming system for MOD (Multimedia on Demand) with EVRC audio codec and UDP-Lite transport protocol. And it performed on mobile phone which has low CPU power and wireless condition. (Especially, this paper focused on audio system only). Through these studies, we show performance improvement of our implemented system compared with classic system. And we discussed about those experimental results.

2. BACKGROUND

2.1 UDP Protocol Over Wireless Network

The problem with UDP is the checksumming policy which will discard packets containing errors. For example, the speech codecs can often handle damaged data but never receive it when using UDP. But, audio/video applications often prefer damaged data over no data since the use of clever codecs can cope for the error. That is to say, classic UDP protocol can not support current error resilient media codecs.

2.2 Concept of UDP-Lite Protocol

The design of UDP-Lite Protocol is centered around the principle of having a partial checksum that covers only the sensitive data which is required to be placed a the beginning of the packet.[5]

Packet Size	Version/Type	Timestamp	Rate series size	Rate series	Extension	EVRC Data
2 byte	2 byte	4 byte	2 byte		2 byte	

Figure 1. Packet Structure

Table 1. UDP Protocol and UDP-Lite Protocol

Source address		
Destination address		
Zero	prot ocol	UDP Length
Source port		Destination Port
Length		Checksum

UDP Protocol

Source address		
Destination address		
zero	prot ocol	UDP Length
Source port		Destination port
Coverage		Checksum

UDP-Lite Protocol

Table 1 shows UDP and UDP-Lite Protocol's header. As the table indicates, the length field in the UDP header is replaced by the coverage field, which signifies how many bytes of the packet have to be checksummed. With a checksum coverage value replacing the packet length, UDP-Lite packets are treated like classic UDP packets with the checksum enabled.[5] So, it is compatible with the existing protocol stack and increases flexibility in systems that can make a use of partially damaged data. Therefore, UDP in this respect becomes a special case of UDP Lite. UDP-Lite Protocol's policy is as below.

- ✓ Errors detected in the sensitive part of the packet - packet gets discarded
- ✓ Errors in the insensitive part of the packet - packet not discarded
- ✓ In summary, UDP-Lite protocol has flexible checksumming schemes that support bit-error resilient codecs. And, provide functionality making it more usable in wireless environments

2.3 EVRC (Enhanced Variable Rate Codec)

The IS-127 Enhanced Variable Rate Codec (EVRC) is based on the Relaxation Code Excited Linear Prediction (RCELP) algorithm.[9] For this high noise suppress ability by noise suppression filter and high voice compress ability, it was adopted to the almost mobile phone these days.

When we use mobile phone, we can not feel any delay. That is reason why EVRC H/W vocoder in mobile phone encodes and decodes voice data very fast. Though sound quality is not quit well, this fast decoding performance makes multimedia (audio) streaming on mobile phone possible. So, in this study, we used EVRC sound codec to stream multimedia data on mobile phone. We can represent EVRC characteristics as follows.

- ✓Input and output of the codec are uniform PCM signals sampled at a rate of 8 kHz, 16 bits/sample
- ✓Encoder produces bit packet no later than 20 ms after it receives last input sample for current frame (160 samples)
- ✓Encoder can generate four different packet types.
  - Rate Value 4 : Rate 1 (171 bits/packet)
  - Rate Value 3 : Rate 1/2 (80 bits/packet)
  - Rate Value 2 : Reserved for future use
  - Rate Value 1 : Rate 1/8 (16 bits/packet)
  - Rate Value 0 : Blank (0 bits/packet)

As stated above, data frame size is 171, when rate value is 4. But, each data frame size is 22byte, 10byte, 2byte in practical for mobile phone use byte alignment to treat input/output PCM data.

3. SYSTEM ARCHITECTURE

3.1 System Overview

Most of mobile phone today uses EVRC en/decoder, which has variable data length by EVRC header rate. Of course, GSM is also used in Europe. But, we focused EVRC in this paper.

For the normal voice telephone call, when mobile phone uses non-transparent mode on RLP (Radio Link Protocol), it drops erroneous packet. But, using transparent mode, it sends erroneous packet to the upper layer instead of retransmitting or dropping. In this study, we tried to recover these erroneous packets by studying EVRC characteristics for making multimedia streaming system on mobile phone. The reason of adoption EVRC audio codec is most of today mobile phone uses EVRC DSP. So, we studied this system using EVRC codec.

If bit error occurs at *Rate* field (EVRC 1byte header), EVRC frame data is useless and may occur error propagation. By this reason, importance of EVRC rate field is so high. And using FEC (Forward Error Correction) has redundancy and needs CPU power too much. So, in this study, *rate series* field that assembles rate values is used to recover rate value error.

3.2 Packet Structure

This implemented system uses packet structure in figure 1 for efficient streaming. *Timestamp* field is 4-byte long, and it presents ms (millisecond). This filed is for playing individual audio and synchronizing audio with video. Today most mobile phone uses msm5100, msm5500 (Qualcomm chipset) which has low CPU power, so *rate series* field consists of individual 1 byte EVRC rate values. Of course, we could assign only 2 bit to rate value for EVRC rate value is between 1 and 4. But, we assigned 1 byte because mobile phone has low CPU power to decode audio and video without H/W support.

In this paper, we mainly focused on audio streaming method. But, real implemented system supports audio, video, and text service. That is to say, we implemented multimedia streaming player on general mobile phone (not smart phone). Of course, to implement this player, we optimized video decoder - we used TCM (Thin Client Media) codec – and avoid bit operation as possible.

Each audio packet has the amount of 400ms EVRC data. So, if we assume that all data is 8 kHz EVRC full rate data (rate 4), packet size is 460-byte long (23-byte/20ms × 400ms). Therefore the whole audio packet size including header is about 500-byte long, and this is appropriate size for fragment in PPP layer on wireless network.

For this packet structure, when packet is sent by UDP-Lite protocol, *rate series* field and *extension* field have to be read, and header part without EVRC data size has to be checksummed. That is to say, *coverage* field in UDP-Lite, which signifies how many bytes of the packet have to be checksummed, is filed with this header size without data size. Therefore UDP-Lite protocol checksums only this header part. Of course, we can insert error correction codes in packet for header part, but header size is relatively small, so probability of error occurrence in header part is also small. If error occurs in this header part, it is proper to retransmit erroneous packet.

The reason of using raw EVRC data, in above *data* field is reason for compatibility. By using this raw EVRC data, packet is able to be decoded by H/W EVRC vocoder and S/W decoder. [8]

4. IMPLEMENTATION PERFORMANCE

4.1 Evaluation Methodology

Multimedia streaming system embedded in mobile phone sometimes makes a tick sound by high error rate on wireless network. Though it uses error resilient codec, erroneous packet is discarded by checksumming in transport, network, or data link layer. So, we used new method, which considers low CPU power and EVRC characteristics as stated above, to deal with this bit errors. And we measured how much the performance of our audio streaming system is improved using UDP-Lite protocol.

To get the real error rate which occurs during sending packets from data server on wire network to mobile phone on CDMA-2000 wireless network, we did various network transmit tests on real wireless network, and then, get the statistics of the error rate. [7]

4.2 Channel Simulation

First of all, we designed and implemented client-server model as figure 3 to simulate wireless network condition. Then, we inserted UDP-Lite simulator, which is able to checksum input packet, into client side. The figure 3 only illustrates audio part without video and text part. But, we implemented all those parts, and our final outcome was media player on mobile phone. And many mobile phone manufacture companies in korea, for example, SAMSUNG, LG, Motorola, etc., adopted this player.

In server part, server encodes PCM data by S/W EVRC encoder and then, makes packet as figure 3. Of course, packet data has to be prepared in real commercial streaming system. But, this system only aim to academic experiment. In client part, client receives packets and reads packet header, then synchronizes A/V/T. In decoding time, client plays EVRC data in *data* field according to rate values in *rate series* field.

The erroneous packets which generated on wireless network have to be transmitted by transparent mode on RLP (Radio Link Protocol), otherwise application on client side can not receive any erroneous packets. To simulate above, we generated errors into data in random, and saved these erroneous data. At this time, error generator in figure 3 generated errors according to actual wireless transmit tests as stated above. The ratio of error generation follows results of table2.

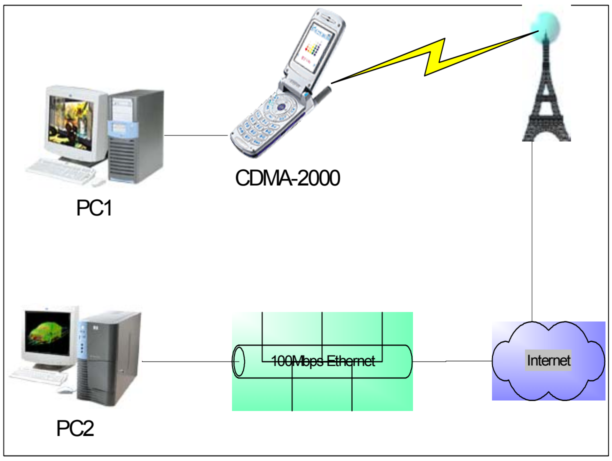

Figure 2. Environment of data transmitting experiment

Figure 2 illustrates an environment of data transmitting experiment on CDMA-2000[7]. By using two PCs and one mobile phone, we gathered many packets to simulate erroneous wireless network. Each time, we changed some factors (e.g. transmitting delay, packet size, etc.) to know characteristics of wireless network. And then, we collected statistics of packet transmitting correctness. By these pre-work, we could test EVRC quality on UDP-Lite Protocol in our simulator.

Table 2. Wire to wireless data transmit experiment results

Packet Size (Byte)	Send Interval (ms)	Receive Interval (ms)	Correct (%)	Send Interval (ms)	Receive Interval (ms)	Correct (%)
300	60	66	92.6	30	35	93.2
350	70	73	95.2	35	40	97.2
400	80	88	86.2	40	50	85.8
450	90	99	92.8	45	59	88.6
500	100	162	82.4	50	55	94.0
550	110	119	93.0	55	65	94.6
600	120	129	92.8	60	70	88.4
650	130	139	86.0	65	83	83.8
700	140	154	89.6	70	79	90.4
750	150	165	91.2	75	95	82.8
1,000	200	209	94.6	100	112	89.0
1,500	300	317	95.4	150	167	90.2
Transmit Speed	40,000 bps			80,000 bps		

Table 2 shows experimental results of data transmit test on CDMA 2000 network. [7]

The way which is proposed in this paper has packets approximately from 450-byte long to 550-byte long, and transmits packet each 400ms. In network testing, correctness had some differences by data transmit interval, but approximately correctness was 90%. And in this experiment, error generator in figure 3 generates errors based on this correctness. Correctness will be high by gathering more traces. With these saved erroneous data, we tested client according to suggested way, and then classic way secondly. This classic way means classic system that has no error resilient technique and uses UDP Protocol.

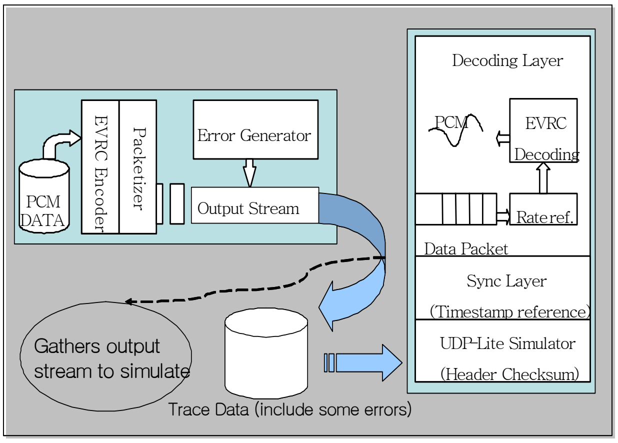

Figure 3. Simulator system architecture

5. EXPERIMENTAL RESULTS & DISCUSSION

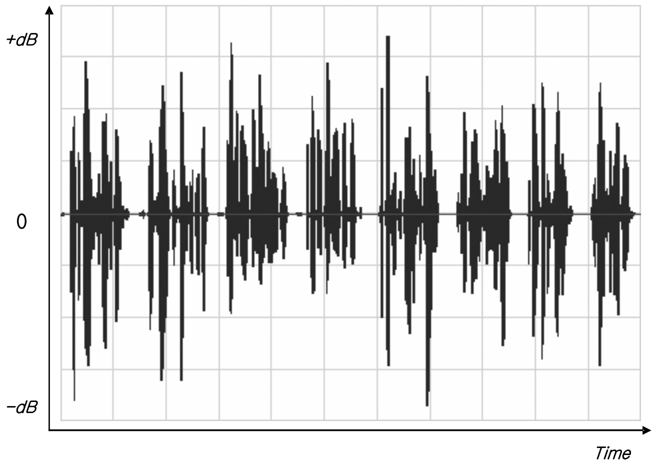

Figure 4-A. Original PCM Data

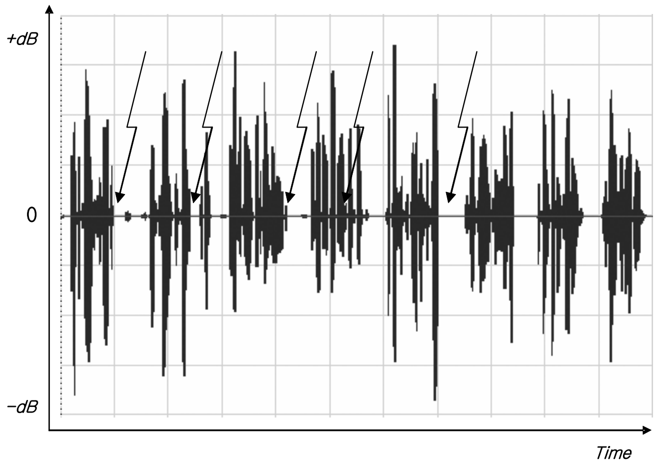

Figure 4-B. Streaming over UDP

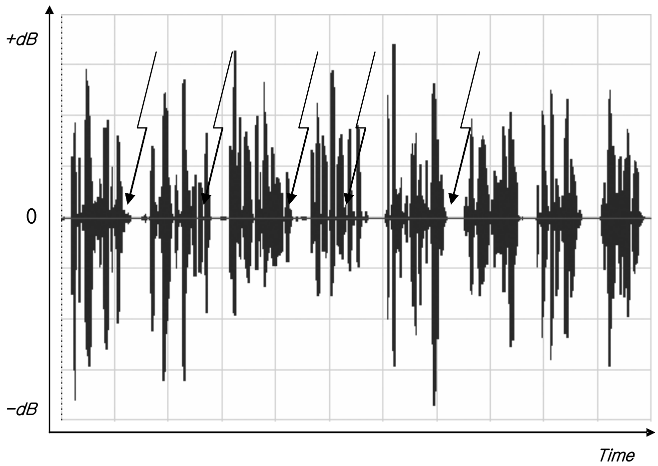

Figure 4-C. Streaming over UDP-Lite

As the figure 4-B indicates, the simulate case of error resilient codec over UDP protocol has some 400ms silent blocks. That is to say, that method shows relatively low sound quality. On the other hand, figure 4-C (streaming over UDP-Lite with algorithm which is proposed in this paper) shows that more similar PCM(Pulse Code Modulation) wave to Original figure 4-A than figure 4-B. and, that is why significant data assigned by coverage field didn't damaged. When we listen to outcome of figure 4-C, we can't notice any difference, likely due to the codec's ability to cope with errors. Small bit difference of encoded EVRC data makes relatively much difference in decoding time. Nevertheless, sound quality in figure 4-C was better than figure 4-B which discard erroneous packets.

To raise performance, we may use noise suppression filter for decoding erroneous packet. When we design system architecture, we also may reduce packet size or guarantee enough buffers and retransmit to cope with an erroneous packet. But, there are a few memories in mobile phone. And mobile phone can not guarantee enough buffers for audio/video streaming together, and moreover, it has many difficult to retransmit packet for high end to end delay on wireless network and non-regular packet arrival time by buffering on base station. Also, we can reduce non-sound block (silent block) by packet size reducing or data interleaving method, but packet size reducing brings many times transmission. And it makes redundant packet header size, then decreases throughput in the end.

Using FEC method increases packet size and need too many CPU power, as stated above, so current mobile phone may not adopt that method. But, using interleaving may prohibit burst error, which is a characteristic on wireless network.

6. RESULTS AND FUTURE WORKS

From above experiments, we found more efficient audio transmitting method, error resilient method, and transport protocol in mobile phone.

For the future, we will try to measure audio quality more objectively (e.g. PESQ), because SNR is inadequate for EVRC. And we will measure network conditions in detail to analyzing wireless characteristics. Especially, transmit audio with video test will be performed according to various policy. By these researches, we will find more efficient multimedia streaming method that is adequate for relatively erroneous wireless condition.

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