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MuxComp –A New Architecture to Improve VoIP Bandwidth Utilization

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***Abstract*—Voice over IP (VoIP) is expected to be the key of the communication industry in near future. Inefficient bandwidth usage by VoIP consumed the network bandwidth. The problem of inefficient bandwidth usage is handled by several techniques such as silence suppression, packets multiplexing, and voice compression. This work proposed new architecture to save network bandwidth for VoIP. We called the new architecture *MuxComp*. *MuxComp*, combines packets multiplexing and voice compression. Whereas, multiplexing the voice packets reduce the overheads, furthermore give the opportunity to compress the packets again. The *MuxComp* architecture consists of two entities. The *MuxCmp* entity is VoIP gateway in the sender side, which performs the multiplexing and compression processes to the packets and sends them to their destinations. The *DCmpDMux* entity is VoIP gateway in the receiver side, which performs the reverse processes of *MuxCmp* through de-compression and de-multiplexing the packets. Furthermore, this work is considered the size of the multiplexed packets, whereas the size must not exceed the network Maximum Transfer Unit (MTU), therefore avoiding the fragmentation delay through the network.**

Keywords-multiplexing; compression; bandwidth utilization; VoIP.

I. INTRODUCTION

The communications world is evolving continuously and quickly. In the last decade Voice over IP (VoIP) occupied a significant situation in communication world, and became one of the main evolving mechanisms in the communication industry. There are many drivers behind the significant of VoIP, especially if we compare VoIP with Public Switched Telephone Network (PSTN). The main driver is the cheapest call rates provided by VoIP, the second driver is the ability to set up other services with voice such as text and video, another vital driver that is VoIP developed rapidly and allow creativity; because of its nature as free and open architecture [1]. Nevertheless, VoIP still suffer from some problems. The main problem is the degrading of QoS (packet loss, delay, and jitter) in comparison of the PSTN. Another important problem is inefficient use of bandwidth [2]. This work will focus on improving bandwidth utilization.

Several techniques are used to improve bandwidth utilization. Here, we will discuss the main three techniques. First technique is silence suppression. The statistical analysis shows that more than 40% of the phone calls are silence. Packetized and send the silence through network waste the bandwidth usage in garbage data, therefore suppress the silence using the suitable mechanism such as Voice Activity Detection (VAD) save the bandwidth greatly. The Second technique is Packets Multiplexing. The size of voice packet payload is usually between 10bytes and 30bytes depend on the codec. The usual 40Bytes VoIP packet header that combined with each packet payload causes header overhead, thus consumes the bandwidth. Hence multiplex multiple payloads in one header reduce the header overhead and save the bandwidth. The third technique is Voice Compression. Compression is the most important technique used to save the bandwidth [3] [4]. Voice compression performed by the codec (compression/decompression). Codec convert the voice from analog to digital, then compress the digital voice using compression algorithm, after that the compressed date converted to frames (packet payload). Frames size varies depending on the codec. Compression ratio varies between codecs depend on the algorithm used by the codec. Table 1 shows the most used voice codecs [5]. This paper proposed novel architecture to improve the bandwidth utilization for VoIP, through combining both, packet multiplexing and voice compression.

The reset of this paper is arranged as follows. Section 2, discussed some of the related works. Section 3, discussed the proposed architecture, showed number of multiplexed payloads, and highlighted how to improve the bandwidth utilization in VoIP by combine packets multiplexing and voice compression. Finally, the conclusion is stated in Section 4.

TABLE I. COMMON VOIP CODEC

Codec	Frame size	Compressed Rate (Bitrate) /kbps
G.723.1 (lr)	30	5.3
G.723.1 (hr)	30	6.3
G.729	10	8
G.729A	10	8
G.729D	10	6.4
G.729E	10	11.8
iLBC (lr)	30	13.33
iLBC (lr)	20	15.2
Speex	20	Various

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II. Related Works

There have been several researches on VoIP bandwidth utilization. Subbiah et al [6] proposed multiplexing of audio streams between IP telephony gateways. The proposed multiplexing method multiplex the audio streams from different users into one RTP payload, which reduces the overhead resulting from combined 40bytes RTP/UDP/IP headers to each audio frame. In order to distinct between the multiplexed frames the method suggested adding mini-header (2bytes) to each multiplexed frame. The result showed that the overhead is reduced by 50% to 80% depends on the audio frame size. Hoshi et al proposed a multiplexing method by combining the RTP packets destined to the same IP telephony gateway into one UDP packet [7]. The proposed method reduces the network load and increases the available bandwidth. After applied this method on H.323 standard, the result showed that the consumed bandwidth decreased 40%, and number of packets decreased to 1/8.

Apart from multiplexing, bandwidth utilization improved through the RTP/UDP/IP header compression. The method of RTP/UDP/IP compression achieves high bandwidth utilization, whereas the method compresses the header from 40bytes to 2 or 4 bytes. Casner and Jacobson rely on two properties to compress the header [8]. First property, most of RTP/UDP/IP header fields are fixed during the communication time. These fixed fields send during the session imitation and eliminated from all other packets. Second property, some of other fields are increased by constant value, using this property the proposed method apply the differential coding to compress these fields.

III. *MuxComp* Architecture

Bandwidth exploitation is critical issue in computer network. Inefficient usage of network bandwidth in VoIP wastes the available network bandwidth. Packets multiplexing and voice compression are two major techniques used to save network bandwidth in VoIP. This work proposed novel architecture to save the bandwidth consumed by VoIP; through combined both packets multiplexing and voice compression, name it ***MuxComp*** architecture. The ***MuxComp*** architecture consists of VoIP gateway in the sender side, and another one in the receiver side. The sender VoIP gateway contains Multiplexer (***Mux***) and Compressor (***Cmp***), name it ***MuxCmp*** VoIP gateway. The receiver VoIP gateway contains De-Compressor (***DCmp***) and De-Multiplexer (***DMux***), name it ***DCmpDMux*** VoIP gateway. The ***MuxCmp*** VoIP gateway multiplex and compress the voice packets and sends them to the ***DCmpDMux*** VoIP gateway. In the other side the ***DCmpDMux*** VoIP gateway decompresses and de-multiplexes the received packets and sends them to their destinations. Fig. 1 clarifies the ***MuxComp*** architecture. We avoid the fragmentation through network, and thus the fragmentation delay, through bind the size of the new packets, after multiplexing and compression, by network MTU.

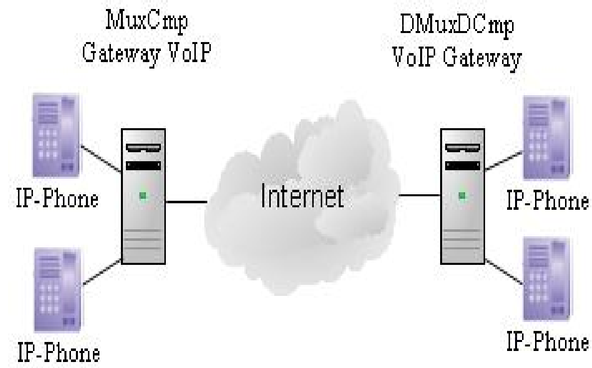

Figure1. MuxComp Architecture

A. Packets Multiplexing

- Number of Multiplexed Payloads.

Packet size affects the network performance. Whereas a big number of small packet size cause overhead, therefore waste the bandwidth. On the other hand, the large packet size requires fragmentation through the network; thus the delay resulting from the fragmentation increases the overall packet delay. Hence, packet size bounded with the network MTU, whereas packet size recommended to be smaller than the network MTU to avoid the fragmentation delay, and big enough to exploit the network MTU [9].

Accordingly, in the ***MuxComp*** architecture the number of multiplexed payloads depends on the network MTU, whereas increase the MTU size increase number of multiplexing payloads, and decrease the MTU size decrease number of multiplexing payloads. Moreover number of multiplexing payloads affected by the compression ratio, whereas the higher compression allows multiplexed additional payloads.

- Packets Multiplexing/De-multiplexing process
  - ***Packets Multiplexing***: the ***MuxCmp*** VoIP gateway receives the packets and check there destinations, after that the ***MuxCmp*** VoIP gateway extracts the packets payload, and the ***Mux*** in the ***MuxCmp*** VoIP gateway multiplex the payloads destined to the same gateway, at last the packet header combined to the multiplexed payloads, and the resulting packets forwarded to their destinations.
  - ***Packets De-Multiplexing***: the ***DCmpDMux*** VoIP gateway receives the packets and extracts the packets payload, then the ***DMux*** in ***DCmpDMux*** VoIP gateway de-multiplexes the payloads, after that the packet header combined with each de-multiplexed part, and the resulting packets forwarded to their destinations. Fig. 2 (a) clarifies the multiplexing and de-multiplexing process.

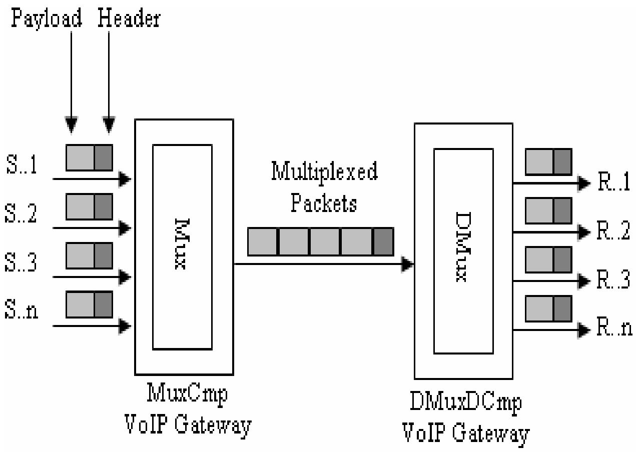

Figure 2 (a). Multiplexing and De-multiplexing process.

B. Voice Compression

- *Improve bandwidth utilization by combine multiplexing and compression.*

Compression algorithms reduce the data size through eliminates the redundant data. The frames (packet payload) generated by the codec are highly compressed by the compression algorithms. We make use of the advantage getting by multiplexing multiple payloads together, since it gives the opportunity to compress the data again which reduce the multiplexed payloads size and save more bandwidth. The compression ratio depends on two factors, first: the compression algorithm. Whereas the compression algorithm must perform high data compression, on the other hand, the compression algorithm must consider acceptable compression time to ensure acceptable overall packets delay. Second: number of multiplexed payloads. Whereas increase number of multiplexed payloads, lead to increases the possibility of repeated data, therefore higher compression [10] [11]. Compression ratio can be calculated using the following equation:

$$CompRatio = \frac{SizeBeforeComp}{SizeAfterComp} * 100\%$$

- Packets compression/De- compression process
  - **Packets Compression:** the ***MuxCmp*** VoIP gateway receives the packets and multiplexes them as in Packets Multiplexing section A, after that ***Cmp*** in the ***MuxCmp*** VoIP gateway compress the multiplexed packets, and the packets forwarded to there destinations**.**
  - **Packets De-compression:** the ***DCmp*** in the ***DCmpDMux*** VoIP gateway de-compress the packets and pass them to the ***DMux***, after that the ***DMux*** de-multiplex the packets and forward them to their destinations as explained in Packets de-Multiplexing section A. Fig. 2 (b) clarify the compression/de-compression process combined with multiplexing/de-multiplexing process.

IV. Conclusion

In this paper, we have proposed new architecture to improve VoIP bandwidth utilization. The ***MuxComp*** architecture combines between two well-known methods used to improve bandwidth utilization, voice compression and packets multiplexing. Voice packets multiplexing give the opportunity to compress the packets again. The ***MuxComp*** architecture consists of two VoIP gateways. First: ***MuxCmp*** VoIP gateway, which multiplexed and compressed the packets. Second: ***DCmpDMux*** VoIP gateway, which de-compressed and de-multiplexed the packets. This work also limits the packets size after multiplexing based on network MTU to avoid the fragmentation delay. The packet size should not be bigger than the smallest MTU through out the path between the 2 ***MuxComp*** entities (***MuxCmp*** VoIP gateway and ***DCmpDMux*** VoIP gateway).

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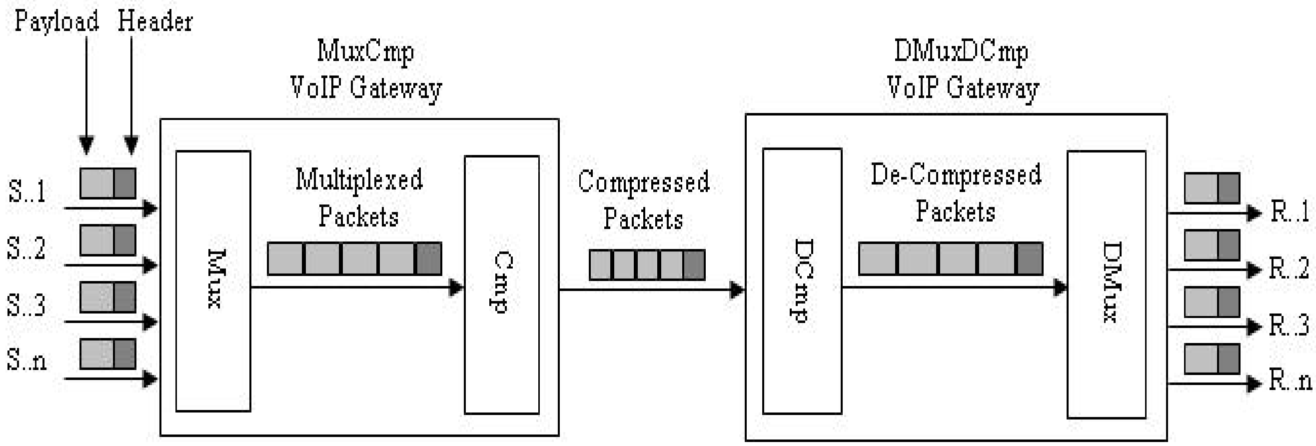

Figure 2 (b). Compression/de-compression process combined with multiplexing/de-multiplexing process.