

## Multiplexing SIP Applications voice Packets between SWVG Gateways

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**Abstract.** SIP applications are expected to dominate the VoIP area in the near future. Unfortunately, the small packet size sending by SIP applications causes packet overhead and network overload. That is, waste the network bandwidth and degrade the voice quality. In order to handle the problems resulting from SIP applications, this paper proposed multiplexing the Voice packets generated by SIP applications. The proposed multiplexing method consists of SIP WAN VoIP gateway (SWVG) in the sender side, and another SWVG gateway in the receiver side. The sender SWVG gateway contains the MUX which multiplexes the packets destined to the same destination. In the other side, the receiver SWVG gateway contains De-MUX which de-multiplexes the received multiplexed packets. This work also handles the delay from multiplexing the packets through binding the packet size by network MTU, thus, avoiding the fragmentation delay. Moreover, a parallel processing for packets multiplexing is proposed to reduce the delay results from the packets multiplexing process.

**Keywords:** Multiplexing; SIP; Bandwidth utilization; VoIP.

### 1. Introduction

Communication technology is one of the most important areas which witnessed noticeable development, in the era of world technology development revolution. In respect to this revolution, the voice communication technology changed from the analog signals carried over by Public Switching Telephone Network (PSTN) to digital packets carried over the internet, this new technology is called voice over IP (VoIP). VoIP exploits the internet infrastructure and protocols to make high quality calls around the world with a cheaper price. Moreover, new VoIP protocols have been developed to enhance the quality and cure-all the communication process. There are several protocols categories used in VoIP systems. Signalling protocols is one of the main protocol categories [1] [2].

Currently, H.323 and Session Initiation Protocols (SIP) are the main standard used as signalling protocols for VoIP. H.323 which were developed by the International Telecommunication Union (ITU), which is complete architecture handle most of the multimedia (audio, video, and data conferencing) communication requirements [3]. Gradually, SIP rollover H.323 and gaining popularity between the network developers. SIP is an Internet Engineering Task Force (IETF) standard defined in RFC 3261. In contrast of H.323, SIP is not complete architecture for multimedia communication. SIP's main purpose is that it initiates and tears down the call session. Accordingly, SIP needs to interoperate with some other protocols to cure-all the communication requirements. One of these protocols is the Real Time Protocol (RTP) which, typically, is used by SIP applications to transmit the voice packets [2] [3] [4].

SIP applications start dominating especially in the VoIP area. Unfortunately, SIP applications exhaust the network bandwidth, since it uses RTP protocol to transfer the voice data. On one hand, the RTP header consists of 12bytes RTP, 8bytes UDP and 20bytes IP with total 40bytes [2] [5]. On the other hand, the packet payload (code frame) usually equal 10, 20, and 30 bytes, depending on the codec. Table1 shows the most

used codec [4]. Typically, this large 40bytes RTP header combined with one codec frame. Hence, the overhead problem is considerable by weighing the RTP header against the packet payload. Moreover, combining the RTP header with one codec frame produces a big amount of small packets. That is, leads to another problem which is overload on the route hops, thus increasing the congestion and packet loss. Thus, degrade the network performance [6]. This work proposed in this paper is to handle the above two problems, packet overhead and network overload, through multiplexing the packets dispatched by SIP applications.

The reset of this paper is organized as follows. Section 2 displays the related works. Section 3 explains the proposed multiplexing method, and then highlights the delay, after that we discuss the proposed method. Lastly, section 4 concludes the paper.

Table1. Most Used Codec

Codec	Frame size	Algorithm Delay	Compressed Rate (Bitrate) /kbps
G.723.1 (lr)	30	37.5	5.3
G.723.1 (hr)	30	37.5	6.3
G.729	10	15	8
G.729A	10	15	8
G.729D	10	15	6.4
G.729E	10	15	11.8
iLBC (lr)	30	30	13.33
iLBC (lr)	20	20	15.2
Speex	20	30	Various
GSM-FR	20	20	5
GSM-HR	20	20	24
GSM-EFR	20	20	18
AMR	20	25	Various

## 2. Related Works

Packet multiplexing has been applied in several environments. Sze et al [7] applied the packet multiplexing on H.323 applications. The proposed multiplexing scheme multiplex several packets from different sources in one packet, this will reduce the overhead resulting from sending large number of small packets, therefore saving the network bandwidth. Moreover, Sze et al, in this work, proposed to compress the RTP header, thus increased the gained bandwidth. By combining these two techniques, packet multiplexing and RTP header compression, the result shows that the bandwidth efficiency increased 300%. On the other hand, Sze et al, consider the delay by binding the multiplexing time by one codec frame period. That is, maintain the overall delay to be within the acceptable delay [7].

Subbiah et al applied the multiplexing method on IP-Telephony Gateways (IP-TG) which connect PSTN/PBX to IP network. Also, the proposed multiplexing method can be applied on IP-TG, which connects cellular access networks with Mobile Switching Center (MSC). That is, proposed method multiplexes multiple VoIP packets from different source in single RTP header. Thus, reduce the overhead resulting from adding 40bytes RTP/UDP/IP header to each codec frame. Moreover, is reduced the number of UDP/IP connections which reduce the congestion on the hops. A 2bytes mini-header is added to each frame in order to distinguishing the multiplexed frames. The result shows that the overhead reduced from 50% to 80% [6].

## 3. Proposed Method

The demand of using VoIP is increasing rapidly and continually, this leads to increasing the number of calls running over the internet, thus, exhausting the network bandwidth which causes bandwidth lacking. There are big efforts from the network developers to handle the bandwidth lacking problem. This work contributes in these efforts through multiplexing the voice packets generated by SIP applications, which improve the bandwidth utilization, and thus reduces the bandwidth lacking. The multiplexing scheme in this work relies on the assumption that there are multiple SIP VoIP LANs connected together via one SWVG gateway, as shown in Fig. 1. That is, increases the traffic on the SWVG gateway, thus, improving the multiplexing process [8]. The proposed multiplexing scheme consists of sender and receiver SWVG gateway. The sender SWVG gateway multiplexes the packets destined to the same destination. The receiver SWVG gateway de-multiplexed the packet and dispatches them to their destinations.

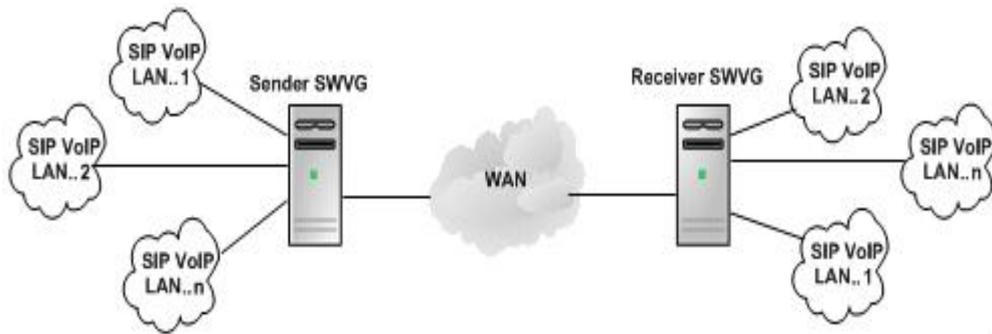


Fig. 1: SWVG Gateway Connect Multiple SIP VoIP LAN

### 3.1. Packet Multiplexing

As mentioned earlier, each codec frame (packet payload) is combined with 40bytes RTP header, which leads to generating and transmitting a big number of small size packets over the network, which intern increase the traffic over the network, which will causes network overload [6]. Moreover, this large header, compared with the codec frames size, causes header overhead.

In order to handle these two problems, network overload and header overhead, this work proposed multiplexing the SIP applications packets between the SWVG gateways. The multiplexing scheme in this work consists of SWVG gateway in the sender side which contains Multiplexer (MUX), and another SWVG gateway in the receiver side which contains De-Multiplexer (De-MUX). The sender SWVG gateway captures the packets dispatched by SIP applications, then it checks the packets destination, after that the MUX in the sender SWVG gateway, multiplex the packets destined to the same SWVG gateway, Fig. 2 (a) illustrate the multiplexing process in the sender SWVG gateway. In the receiving side, the receiver SWVG gateway de-multiplexes the received packets, and then dispatches the packets to their destinations, Fig. 2 (b) illustrates the de-multiplexing process in the receiver SWVG gateway.

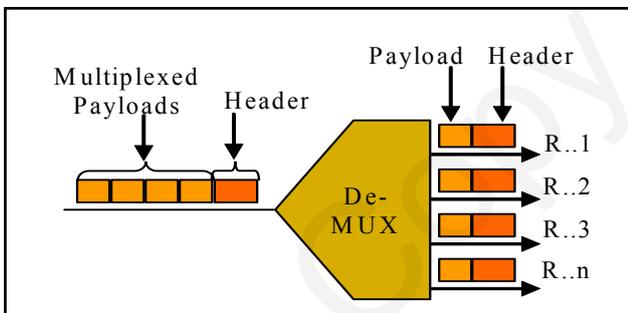


Fig. 2 (a): Packets Multiplexer

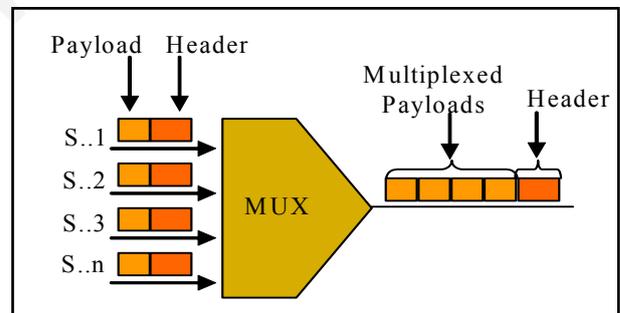


Fig. 2 (b): Packets De-Multiplexer

### 3.2. Delay

Delay is the main issue in VoIP. Unfortunately, packet multiplexing affects the VoIP delay since it increases the packet size, whereas, the large packet size requires fragmentation through the network, which causes additional delay. In order to handle the fragmentation delay, the multiplexed packet size bounded by the network Maximum Transfer Unit (MTU) [5].

Moreover, packet multiplexing process produces additional delay on the SWVG. Whereas, the SWVG captures and processes the SIP application packets, and classifies the packet according to there destinations, and then process and multiplex the packets. The packet processing and multiplexing delay can be reducing by using parallel processing for multiplexing packet going to different destinations.

Furthermore, the multiplexing period affects the delay. Therefore, increasing the multiplexing period, increase the delay, along with increasing the number of multiplexing packets, hence decrease the overhead. On the other hand, decreasing the multiplexing period decreasing the delay, and decrease the multiplexing packets, thus increasing the overhead. Accordingly, the multiplexing period must compromise between the delay time and the header overhead [8].

### 3.3. Method discussion

Typically, SIP applications used RTP protocol to transmit the voice packet over the network. The overhead resulting from the RTP header waste the network bandwidth. The ratio of waste bandwidth (packet overhead) depends on codec frame size. Packet overhead can be calculated using equation1 below:

$$Po = (hs / Ps) * 100\% \quad (1).$$

Where Po is the packet overhead, hs is the packet header size which equal 40bytes, Ps is the packet size which depends on the codec frame size plus the 40bytes header size.

By refer to equation1 and frame size in table1, the typical overhead can be calculated as follows:

- 1) When the codec frame size is 10Bytes: overhead ratio is  $(40 / (10+40)) * 100\% = 80\%$ .
- 2) When the codec frame size is 20Bytes: overhead ratio is  $(40 / (20+40)) * 100\% = 67\%$ .
- 3) When the codec frame size is 30Bytes: overhead ratio is  $(40 / (30+40)) * 100\% = 57\%$ .

The waste bandwidth is obvious in all cases. By multiplexing the packets, on the SWVG gateway, the ratio of waste bandwidth will be reduced. The ratio of reduced overhead depends on the number of multiplexing packets (Multiplexed packet size). Whereas, increasing the number of multiplexing packets decrease the overhead and vice versa. Unfortunately, the number of multiplexing packets is restricted by two factors. The first one is the network MTU, the big MTU size increases the number of multiplexing packets, and small MTU size decrease number of multiplexing packets. The second one is the multiplexing period, the multiplexing period should keep the delay time within the acceptable delay time. The following algorithm depicts how the MTU and multiplexing time restrict the number of multiplexing packets.

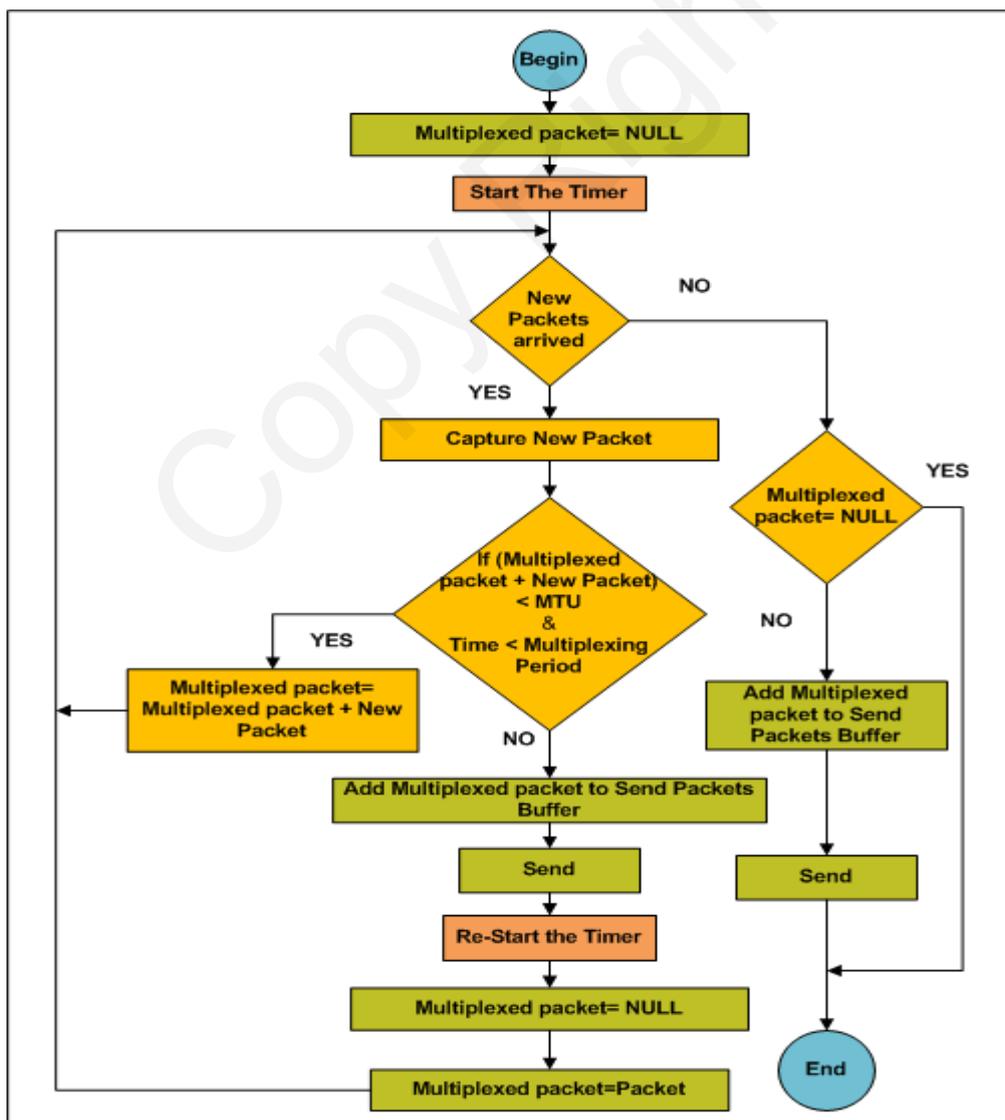


Fig. 3: Packet Size Restricted by MTU and Multiplexing Period

Moreover, multiplexing the packets reduce the number of sending packets, thus decreasing the network overload, therefore, reducing the packet loss and congestion which improve the Quality of service [6].

#### 4. Conclusion

In this paper, we have proposed multiplexing method that multiplex the packets generated by SIP applications. This method will, reduce the header overhead, thus saving the network bandwidth, moreover, reducing the number of sending packets, thus reducing the overload on the network hops. On the other hand, we have considered the delay resulting from the multiplexing process, through binding the multiplexed packets size by the network MTU, thus avoiding the fragmentation delay. Moreover, we have used a parallel processing to achieve the multiplexing process, thus reducing the processing delay.

#### 5. References

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