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A NEW METHOD TO BOOST VOIP PERFORMANCE OVER IPV6 NETWORKS

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The demands on virtual communication have increased noticeably during the COVID-19 pandemic lockdown. As an essential part of virtual communication, VoIP should be promoted to achieve the desired level of performance. One critical area investigated in VoIP is the bandwidth utilization (BWU) of the VoIP over IPv6 networks. Enhancing BWU will impact call capacity and quality; it increases call capacity and boosts call quality. Unfortunately, a considerable amount of bandwidth is wasted when running VoIP over IPv6 networks. This is due to the large size of the IPv6 header and the small size of the speech frame. This paper proposes a new method to handle the inefficient BWU when running VoIP over IPv6 networks. The proposed method combines multiplexing multiple VoIP packets in one IPv6 header and using the superfluous fields to carry a portion of the speech frame. Therefore, the proposed is called packet multiplexing and carrier fields (PMCF). Investigation of the PMCF method has been done using four metrics to measure the promotion in BWU, namely calls capacity, header size, bandwidth saving, and speech frame shortening metrics. With the four metrics, the PMCF method has outperformed the comparable methods. For instance, the call capacity has been promoted by up to 269% compared to the typical IPv6 method in the tested scenarios. Therefore, the PMCF method is a feasible solution to facilitate the BWU of VoIP when running VoIP over IPv6 networks.

Keywords: VoIP, bandwidth utilization, IPv6, VoIP performance

1. Introduction

At the beginning of 2020, the demand for virtual communications has increased dramatically and rapidly due to the COVID-19 pandemic lockdown. People couldn't meet but needed to communicate for various purposes such as work, education, and even medication (Raycheva *et al.*, 2020; Garro_Abarca *et al.*, 2021). The real-time voice and video over IP have served these purposes to a great degree. As for VoIP (voice over IP), the forecasting for 2021 is around 3 billion subscribers (Hoffman, 2019; Abualhaj *et al.*, 2020a). Therefore, VoIP networks should be enhanced to carry out the calls thoroughly. One of the metrics that should be investigated to enhance the performance of the VoIP networks is bandwidth utilization (BWU). Improving BWU is directly proportional to the call's capacity and quality; it increases call capacity and boosts the call quality (Abualhaj *et al.*, 2020a).

The nature of VoIP traffic is highly depressing the BWU of VoIP networks because the VoIP packet header size is immense compared to the VoIP packet payload. In many cases, the VoIP packet header is larger than the VoIP packet payload. On the one hand, the VoIP packet payload is constructed from one or more digital speech samples produced by a codec. Codecs are making very small speech samples, typically from 10 to 30 bytes. Increasing the speech sample size will increase the latency of the VoIP, which is not acceptable for the call quality. Therefore, the speech sample should be small and thus the VoIP packet payload. For instance, the G.729 codec produces a 10-byte speech sample as a packet payload. Table 1 shows the typical length of the speech sample of the most common codecs (Tomoskozi *et al.*, 2016; Abualhaj *et al.*, 2021a; Gupta *et al.*, 2018).

On the other hand, the VoIP packet header is constructed from three protocols, excluding the layer 2 protocol, namely RTP, UDP, and IPv4/IPv6. In IPv4, the header's length (RTP/UDP/IPv4) is 40-byte, which consumes around 80% of the total bandwidth with the G.729 codec. The consumed bandwidth is even more in the case of IPv6 protocol. The RTP/UDP/IPv6 header length is 60-byte, consuming 85.7% of the total bandwidth with the same G.729 codec (Abualhaj *et al.*, 2021a; Narayan, 2010). Table 2 shows the consumed BW when using the IPv6 protocol with different codecs. As we can see, the packet's header consumed an excessive amount of bandwidth, especially when using IPv6 protocol, and depress the quality of the calls. Several researchers have investigated this issue. In this study, further investigation of this issue will be introduced, and a new technique will be designed to make this issue less severe.

The remaining part of the paper has the following structure: the related works in Section 2 covers some of the main multiplexing methods. The proposed method to handle the BWU issue of VoIP is presented in Section 3. Section 4 investigates the proposed method in comparison to the other method in terms of BWU. Section 5 presents the finding and conclusion of the paper.

Table 1. Common speech codecs

Codec	Sample Size (byte)	Mean Opinion Score (MOS)
G.729	10	4.1
G.728	10	3.61
G.723.1	24	3.9
G.726	20	3.85

Table 2. Consumed BW by packet header

Codec	Protocol	Common Payload size	Packet size	Header Consumed BW
G.729	IPv4	10	50	80%
	IPv6	10	70	85.7%
G.728	IPv4	10	50	80%
	IPv6	10	70	85.7%
G.723.1	IPv4	24	64	62.5%
	IPv6	24	84	71.4%
G.726	IPv4	20	60	66%
	IPv6	20	80	75%

2. Related Works

BWU is a critical issue in the VoIP sector because it affects both call capacity and call quality. Many types of research with different approaches have been suggested to alleviate the severity of poor BWU. One of the top approaches is packet multiplexing, which proposing piggybacking many small VoIP packets in one IP header (RTP/UDP/IP) (Abualhaj et al., 2020b; Seytnazarov and Kim, 2017; Sze *et al.*, 2002; Casner and Jacobson, 1999). For elaboration, assume the first packet is transmitted to the destination D. Then, theoretically, all packets transmitted to destination D through the same route will be piggybacked in the header of the first packet. In general, the packets are piggybacked up to a certain length, period, or other critical VoIP performance metric. Figure 1 elaborates the packets multiplexing approach. In the figure (Figure 1), the NPkt refers to the ordinary VoIP packet, the packet before the multiplexing process, and without any modification. In the same figure (Figure 1), the MPkt refers to the multiplexed VoIP packet, resulting from piggybacking several NPkt packets in one header.

Figure 1. Packet multiplexing approach

Many methods were designed under the multiplexing approach. One of the first methods was developed in Nokia Research Center by Subbiah *et al.* (Subbiah, 1999). Speech payload from different NPkt packets is piggybacked into one MPkt packet. The multiplexing occurs at the application layer of the OSI stack, in which multiple speech payloads are piggybacked into one RTP/UDP/IP header. A 2-byte header is placed at the beginning of every speech payload to separate the piggybacked speech payloads. The proposed method is designed to work on the VoIP gateways that connect the PSTN/PBX (Public Switched Telephone Network/Private Branch Exchange) systems and cellular access networks. The

investigation result showed that the BWU of VoIP networks has promoted up to 80%. Besides, the overcrowding in intermediary network devices and processing latency in MPLS (Multiprotocol Label Switching), IPsec, and diff-serv networks are alleviated.

More recently, Abu-Alhaj *et al.* proposed a multiplexing method called Delta-Multiplexing (Abualhaj *et al.*, 2010). Similar to the previous method, the multiplexing process in Delta-Multiplexing occurs at the transport layer of the OSI stack, in which multiple speech payloads and an RTP header for each payload are piggybacked into one UDP/IP header. Besides the multiplexing, the Delta-Multiplexing method applies a delta algorithm at the speech payload. That is, the delta algorithm finds the difference between the successive packets' speech payload. Only the difference between the speech payload is sent rather than sending the full-length speech payload. The delta algorithm performs the opposite operations at the receiver side to construct the original full-length speech payload. A 2-byte header is placed at the beginning of every mini-packets (RTP header + speech payload) within the MPkt packet to reconstruct the NPkt packets at the receiver side. The multiplexing process and delta algorithm have promoted BWU of VoIP networks between 68% and 72%.

One of the best multiplexing methods was invented and patented by Roay under the auspices of Cisco Technology Inc. (Roay, 2013). The multiplexing occurs at the network layer of the OSI stack, in which multiple mini-packets (RTP/UDP header + speech payload) are piggybacked into one IP header (IPv4 or IPv6). Only the first mini-packet within the MPkt packet has a flag that indicates that this is an MPkt packet. This flag is used by the receiver gateway to process the incoming packet as an NPkt packet or an MPkt packet. The ability of the receiving gateway to process the MPkt packet is negotiated through call establishment using either SIP or H.323 signaling protocols. If the receiving gateway cannot process the MPkt packet, all the packets to that gateway must be NPkt packets. The MPkt packet does not wait for the NPkt packet to be available but rather piggybacked only the NPkt packets in the queue. This is to avoid imposing an additional delay that may impact the call quality. If the MPkt packet reaches the MTU, it is sent to the destination, even if more NPkt packets are in the queue. The saved bandwidth is 20-byte in IPv4 or 40-byte in IPv6 for every NPkt packet within the MPkt packet.

All the previous works are proposing a multiplexing method to promote the BWU. The method proposed by Roay is designed to be compatible with the current environment since it integrates the RTP/UDP header in every mini-packet. In addition, unlike the other methods, it does not impose any extra delay that impacts the quality of the calls. Moreover, Roay method is the only, up to my knowledge, that discussed and considered IPv6 networks. This makes it a suitable multiplexing method for the delay-sensitive VoIP applications over IPv6 networks. Though, adding the RTP/UDP header to every mini-packet still degrades the BWU severely. Accordingly, this paper adopts the same method proposed by Roay. Still, it uses the superfluous fields in the headers to carry the speech payload of a packet. This will keep the features of the Roay method and promotes the BWU. The proposed is called packet multiplexing and carrier fields (PMCF). The PMCF method will be designed to work over IPv6 networks.

3. PMCF Method

The PMCF method is destined to promote the BWU when running VoIP over IPv6 networks. Multiplexing more VoIP packets in one MPkt packet will help to maximize the BWU. Therefore, the PMCF method should be deployed in a network location where sufficient VoIP packets are passed. This location might be the exit point of a LAN with many VoIP clients or even the exit point of several LANs, each of them with many VoIP clients. In Figure 1, the Netwok1 gateway is the location at which the PMCF method could be implemented. The PMCF method executes two primary mechanisms: the PMCF sender side (PMCFs) and PMCF receiver side (PMCFr). The PMCFs and PMCFr mechanisms will be explained in Subsection 3.2 and 3.3, respectively.

3.1. Superfluous Fields

The VoIP packet consists of three headers: RTP, UDP, and IPv4/IPv6 (IPv6 is the focus of this work). These 60-byte RTP/UDP/IPv6 headers contain much information that serves diverse applications other than VoIP. Only part of this information is necessary to convey the speech payload to the intended destination. The other part of the RTP/UDP/IPv6 headers information is superfluous for VoIP applications and used by other applications. In VoIP applications, this information is conveyed with every packet meaninglessly, creating an unfavorable reduction of BWU (Abualhaj *et al.*, 2021b; Tomsho, 2019; Hartpence, 2013; Perkins, 2003; Gao *et al.*, 2019). The PMCF method will turn the headers' fields that carry this information to carrier fields for the speech payload itself. This way, the speech payload is shortened, and the BWU is promoted.

The first header of the VoIP packet is the RTP, which precisely conveys the real-time data in all scenarios. Though part of the information in RTP is not used to represent the speech payload in specific scenarios, it will be used in other scenarios. Therefore, the information in the RTP header should stay as is to make sure that the PMCF method works in all VoIP scenarios (Hartpence, 2013; Perkins, 2003). The second header of the VoIP packet is the UDP, which is used with various applications other than real-time applications. Some of the information in the UDP header is optional and disabled (set to zeros) for many applications, including the VoIP applications, namely the Checksum and the Source Port. Therefore, the fields that carry the Checksum and the Source Port information will be used by the PMCF method as carrier fields for the speech payload (Tomsho 2019; Eggert *et al.*, 2017; Khan *et al.*, 2016; Stein *et al.*, 2020). The third header of the VoIP packet is the IPv6, which is used with all applications in the IP-based networks. VoIP applications do not need the Source IPv6 address information because VoIP applications are not "request/response" applications. However, there is only one IPv6 header with one Source IPv6 address field to the entire MPkt. Therefore, it is not worth using the Source IPv6 address field as a carrier field to the speech payload. In addition, there are several NPkt from different sources within the MPkt. Hence, which of the NPkt's speech payload will be carried by the Source IPv6 address field. This will complicate the process with no real benefit.

In a nutshell, the Checksum and the Source Port fields in the UDP header will be used by the PMCF method as carrier fields for the speech payload. All the other information in the RTP/UDP/IPv6 headers is needed to convey the speech payload in different environments and scenarios. The Checksum and the Source Port fields are part of every mini-packet (RTP + UDP + speech payload) within the MPkt. Therefore, 4-byte of the speech payload of every mini-packet will be truncated. Figure 2 shows the mini-packet of the PMCF method.

Figure 2. Mini-packet format

3.2. The PMCFs Mechanism

The PMCFs mechanism piggybacked several NPkt packets in one sizeable MPkt packet at the sender side. For each NPkt packet in the waiting queue, the mini-packet is extracted. Then, 4-byte of the speech payload of every mini-packet is placed in the Checksum and the Source Port fields of the mini-packet header (RTP + UDP). This produces mini-packets with a smaller speech payload. After that, the produced mini-packets destined to the same destination will be piggybacked in one IPv6 header. Figure 3 shows the format of the MPkt packet. When the size reaches the MTU, or the waiting queue is empty of the VoIP packets, then the MPkt packet is transmitted to the destination. The destination IPv6 address of each call is stored in a particular table on both sender and receiver sides. This table is called the destination IPv6 address (Dst IPv6) table. The purpose of the Dst IPv6 table (Table3) is discussed in Subsection 3.4. Figure 4 shows the flowchart of the PMCFs mechanism.

Figure 3. MPkt packet format

Figure 4. The PMCFs mechanism

3.3. The PMCFR Mechanism

The PMCFR mechanism segregates the MPkt packet to restore the original NPkt packets. At first, the IPv6 header is stripped out from the MPkt packet. Then, the mini-packets within the MPkt payload are separated. This is done based on the Length field in the UDP header of each mini-packet. Next, the speech data inside the Checksum and the Source Port fields are retrieved and placed at the end of the speech payload of the mini-packet. This constructs the mini-packet with the original full-length speech payload. Both Checksum and the Source Port fields are disabled to avoid misinterpreting them by the callee client. After that, the IPv6 header is attached to each mini-packet to construct the original NPkt packet. The destination IPv6 address of each NPkt packet is restored from the Dst IPv6 table (Table 3), as discussed in Subsection 3.4. Finally, the NPkt packet is sent to the callee client. Figure 5 shows the flowchart of the PMCFR mechanism.

Figure 5. The PMCFR mechanism

3.4. The Dst IPv6 Table

The PMCFR mechanism adds the IPv6 header to the mini-packet, including the destination IPv6 address to construct the NPkt packet. The PMCFR method keeps the destination IPv6 address in the Dst IPv6 table at both sender and receiver sides. A flag is needed to identify the destination IPv6 address of a specific call's packets. The PMCFR mechanism uses the SSRC in the RTP header for this purpose. The SSRC is a unique value in the RTP assigned for each call (Abualhaj *et al.*, 2020b; Perkins, 2003). The SSRC is stored in the Dst IPv6 table along with its corresponding destination IPv6 address. When segregating the MPkt packet, the SSRC in the RTP of the mini-packet header is used to find the corresponding destination IPv6 address from the Dst IPv6 table. In sporadic cases, the SSRC of two calls could be matched. In these cases, the next available SSRC value is used. Table 3 exhibits the Dst IPv6 table at PMCFR and PMCFR.

Table 3. Dst IPv6 Table

PACFS			PACHR	
SSRC in RTP	SSRC in Dst IPv6 Table	Destination IPv6	SSRC	Destination IPv6
126593730	126593730	2001:1234::1a13	126593730	2001:1234::1a13
126593731	126593731	2001:1234::1a14	126593731	2001:1234::1a14
126595730	126595730	2001:1234::1a15	126595730	2001:1234::1a15
126595730	126595731*	2001:1234::1a16	126595731	2001:1234::1a16
126593730	126593732*	2001:1234::1a17	126593732	2001:1234::1a17

*Next available SSRC value

4. PMCF Investigation

This section investigates the impact of the PMCF on BWU when running VoIP over IPv6 networks. The PMCF was investigated compared to the typical IPv6 method (without multiplexing or carrier fields) and Roay method (Roay, 2013). The BWU improvement of the three methods was investigated in terms of call capacity, header size, bandwidth saving, and speech frame shortening metrics. The suggested PMCF technique along with the comparable methods were simulated on the well-known Network Simulation 3 (NS3). The simulation topology includes two components of the PMCF technique, PMCFs and PMCFR, which discussed in sections 3. The simulation scenario assumes the PMCFs at the network exit-router at the sender side and the PMCFR at the network enter-router at the receiver side. The link between the two routers is represented as a buffer of 100 packets. Ten different experiments with different link bandwidths were used to compare the three methods using the metrics above. The link bandwidth started at 100kb and increased by 100kb for each different experiment up to 1000kb. Each of these experiments was run with two codecs, namely G.728 and G.726. A CBR (constant bit rate) traffic generator is used with each end.

4.1. Calls Capacity

Figures 6 and 7 display the call capacity of the PMCF method compared to the IPv6 method and Roay method, using G.26 and G.28 codecs, respectively. The PMCF method showed the highest call capacity with the two codecs compared to the IPv6 method and Roay method. In addition, the call capacity with G.28 codec has been promoted better than G.26 codec. For instance, the number of calls with the G.28 codec at 1000kb bandwidth is 94, 81, and 35 when running the PMCF method, Roay method, and IPv6 method, respectively. Therefore, the call capacity is promoted by up to 269% and 231% over the typical IPv6 method, when using the PMCF method and Roay method, respectively.

On the other hand, the number of calls with the G.26 codec at 1000kb bandwidth is 68, 61, and 31 when running the PMCF method, Roay method, and IPv6 method, respectively. Therefore, the call capacity is promoted by up to 219% and 197% over the typical IPv6 method when using the PMCF method and Roay method, respectively. The apparent reason for the call's capacity promotion is piggybacking several NPkt packets in one MPkt packet with one IPv6 header rather than a dedicated IPv6 header to each NPkt. However, the G.28 codec shows higher call capacity promotion than G.26 codec because the speech payload of the G.26 codec is 20-bytes while the speech payload of the G.28 codec is 10-bytes. Thus, placing 4-byte of speech payload in the carrier fields will impact the 10-byte G.28 codec more than the 20-byte G.26 codec. Therefore, the smaller the speech payload of a codec, the higher the call capacity is promoted when using the PMCF method.

Figure 6. Call capacity (G.26 codec)

Figure 7. Call capacity (G.28 codec)

4.2. Header Size

Figure 8 displays the header size ratio of the PMCF method compared to the IPv6 method and Roay method using G.26 codec. The PMCF method showed the lowest header size ratio compared to the IPv6 method and Roay method. For instance, the header size ratio at 100 calls is 45%, 51%, and 75% when running the PMCF method, Roay method, and IPv6 method, respectively. Therefore, the wasted bandwidth when running the PMCF method is less than the other comparable methods. This is due to piggybacking several NPkt packets in one MPkt packet with one IPv6 header rather than a dedicated IPv6 header to each NPkt. In addition, the 4-byte Checksum and the Source Port carrier fields are not considered header fields because they are carrying the part of the speech samples.

Figure 8. Header size ratio

4.3. Bandwidth Saving

Figures 9 and 10 display the bandwidth saving of the PMCF method and Roay method compared to the traditional IPv6 method, using G.26 and G.28 codecs, respectively. Again, with the two codecs, the

PMCF method showed higher bandwidth saving than Roay method. In addition, the bandwidth saving with G.28 codec has promoted better than G.26 codec. For instance, the bandwidth saving with the G.28 codec at 100 calls is 62% and 57% when running the PMCF method and Roay method compared to the IPv6 method, respectively. Therefore, the PMCF method showed better bandwidth saving by up to 5% than the Roay method. On the other hand, the PMCF method showed bandwidth saving with the G.26 codec equals 55% at 100 calls which is less than the G.28 codec. The apparent reasons for the bandwidth saving are the exact reasons for promoting the capacity of the call.

Figure 9. Bandwidth saving (G.26 codec)

Figure 10. Bandwidth saving (G.28 codec)

4.4. Speech Frame Shortening

Figure 11 displays the speech frame shortening ratio when using the PMCF method. The shortening ratio is 20% and 40% when using the G.26 codec and G.28 codec, respectively. The resulting shortening is due to place part of the speech frame in the 4-byte carrier fields. In addition, the shortening ratio is more when using the G.28 codec compared to the G.26 codec. This is because the speech frame of the G.28 codec is smaller than the speech frame of the G.26 codec. Therefore, the smaller the speech payload of a codec, the higher the shortening ratio.

Figure 11. Speech frame shortening ratio

5. Conclusion

VoIP has played a crucial role in virtual communication during the COVID-19 pandemic lockdown. Unfortunately, the size of the VoIP packet header, particularly over IPv6 networks, is consuming most of the bandwidth, with little bandwidth remaining for the actual speech data. The paper has proposed the PCMF method to promote the BWU of VoIP over IPv6 networks. The PCMF method piggybacked many IPv6 VoIP packets in one IPv6 header. It also uses the Checksum and Source Port fields in the UDP header of each mini-packet to carry a portion of the speech frame. These two techniques have promoted the BWU considerably. For instance, the call capacity has been boosted by up to 269% when using the G.28 codec at 1000kb network bandwidth. In addition, the bandwidth saving with the same G.28 codec at 100 calls is 62% of the available bandwidth compared to the IPv6 method. This promotion of the BWU will also boost the quality of the calls. Therefore, the PCMF method can work successfully to promote VoIP's overall performance over IPv6 networks. As future works, the impact of the PCMF method on the VoIP security will be investigated.

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