BULGARIAN ACADEMY OF SCIENCES

CYBERNETICS AND INFORMATION TECHNOLOGIES • Volume **20**, No 3
Sofia • 2020 Print ISSN: 1311-9702; Online ISSN: 1314-4081
DOI: 10.2478/cait-2020-0034

PS-PC: An Effective Method to Improve VoIP Technology Bandwidth Utilization over ITTP Protocol

Mosleh M. Abualhaj[1], *Sumaya N. Al-Khatib*[2], *Qusai Y. Shambour*[2]

[1]*Department of Networks and Security Information, Faculty of Information Technology, Al-Ahliyya Amman University, Ammann, Jordan*
[2]*Department of Software Engineering, Faculty of Information Technology, Al-Ahliyya Amman University, Ammann, Jordan*
E-mails: m.abualhaj@ammanu.edu.jo sumayakh@ammanu.edu.jo q.shambour@ammanu.edu.jo

Abstract**:** *Voice over IP* (*VoIP*) *wastes a valuable amount of bandwidth because of its large packet header size compared to its small packet payload. The main objective of this paper is to reduce the amount of this wasted bandwidth, by proposing a new packets coalescence method, called Payload Shrinking and Packets Coalesce* (*PS-PC*)*. The proposed PS-PC method reduces the amount of the wasted bandwidth by i) coalesces a group of VoIP packets in one header instead of a separate header to each packet and ii) shrinks the VoIP packet payload to a smaller one based on a certain algorithm. The proposed PS-PC method is deployed at the sender side VoIP gateway that represents an exit point to a myriad number of simultaneous VoIP calls. The performance evaluation showed better bandwidth usage when deploying the proposed PS-PC method with ITTP protocol in comparison to the traditional ITTP protocol without the PS-PC method.*

Keywords**:** *ITTP, VoIP Protocols, Packet Coalesce, Header Overhead, Bandwidth Utilization.*

1. Introduction

Internet is everywhere; Internet technologies are everywhere. One of the increasingly deployed Internet technology is Voice over Internet Protocol (VoIP). There is a demand for VoIP in just about every modern business, educational institution or government organization [1, 2]. The main reason for the high demand on VoIP is the very low or free call cost [3]. This growing demand has brought the attention of VoIP developers and researchers. They have concentrated on the two main issues that face the VoIP to replace the current traditional phone systems; these are VoIP QoS and VoIP bandwidth utilization [4, 5]. As for VoIP bandwidth utilization, the three main approaches to make it more efficient are packet header compression, packet payload compaction, and packets coalescence [5-7]. The "packet payload compaction" approach has successfully compacted the typical 40 bytes VoIP packet header to 2 or 4 bytes [7]. The packet payload compaction approach uses a hardware/software called a codec to compact the voice data after converting it from analog to digital. It also uses "voice activity detection" techniques to suppress silence [6, 8]. The packets coalescence approach coalesces a group of VoIP packets in one header instead of a separate header to each packet. It inspects the destination of the VoIP packet, separates the packet header from the packet payload, and coalesces the packets that are travelling to the same destination in one header [3, 5].

VoIP uses several protocols to carry the voice call over the Internet. These protocols are divided into two main groups, based on their functions, the signaling protocols group and the voice data transfer group [5, 9]. H.323 and SIP are the two standard signaling protocols, which are used to setup the call between the caller and callee [9, 10]. The voice data transfer protocols are the 20-bytes Real-time Transport Protocol (RTP), the 12-bytes Inter-Asterisk eXchange (IAX), and the 6-bytes Internet Telephony Transport Protocol (ITTP). These protocols are used to carry the voice data between the caller and callee after call setup. Both IAX and RTP protocols work in conjunction with UDP protocol in order to be able to carry the voice data while the ITTP protocol is able to carry the voice data by itself [11-13]. Fig. 1 shows the typical ITTP packet format.

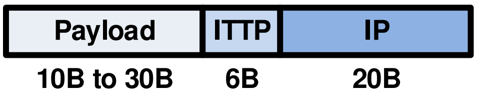

Fig. 1. ITTP packet format

VoIP codecs are typically producing small VoIP packet payloads between 10 to 30 bytes [14]. Appending 26 bytes ITTP/IP VoIP packet header to this small payload causes extreme header overhead and thus high wasting of bandwidth [15]. As mentioned earlier, VoIP packet coalescence methods are used to group several VoIP packets payload in one header to lessen the header overhead and improve the bandwidth utilization. In this research, we will suggest a new packets coalesce method to improve bandwidth utilization of VoIP when using ITTP protocol. In addition, the suggested method will integrate a new mechanism that compact the VoIP packet payload.

The rest of this research is organized as follows. Section 2 briefly summarizes some previous works on packet coalesces and packet payload compaction. Section 3 elaborates the proposed method components and algorithms. Section 4 shows the performance evaluation of the proposed method in comparison to the traditional ITTP method. Section 5 concludes the paper.

2. Related works

Over the past years, many VoIP researchers were focusing on improving bandwidth utilization across various approaches. In this section, we briefly summarize related works of this article on packet coalesces and packet payload compaction.

In 2014, Vulkan et al. [16] have suggested a VoIP packets coalesce method to improve bandwidth utilization. The suggested method reduces the overhead of the small size VoIP packets by coalesces several packets that are belonging to different

VoIP streams in one UDP/IP header. The suggested method can be deployed in different environments, in which there is a myriad number of VoIP packets, such as a mobile system environment where user equipment is attached to the system via an I-HSPA Node. According to the suggested method, the VoIP packets are coalescing for a predefined period of time or until the size of the coalesced packet reaches the maximum transmission unit (e.g., 1500 bytes in case of Ethernet). Clearly, these values can be changed based on the environment in which the suggested method is deployed [16]. In 2015 the authors have suggested a VoIP packets coalesces method to improve bandwidth utilization over ITTP protocol, named ITTP-Multiplexing (ITTP-Mux). The ITTP-Mux works in environments that have several VoIP LAN networks that are sharing the same VoIP WAN gateway. In such an environment, there will be plenty of VoIP packets for the ITTP-Mux method to be efficient and improves bandwidth utilization. The VoIP WAN gateway that is located at the transmitter side performs the packets coalesces while the VoIP WAN gateway, located at the destination side performs packet segregation (separate the coalesced packet to the original packets). The coalesces process occurs at the ITTP layer, in which several packets coalesce in one ITTP/IP header. One-byte mini-header is attached to each mini-packet within the coalesced packet. The purpose of this mini-header is to distinguish the mini-packets within the coalesced packet at the destination side VoIP WAN gateway. The performance evaluation of the ITTP-Mux showed that the bandwidth utilization has enhanced by up to 29.1% in comparison to the traditional method [15].

Apart from packets coalesces methods, VoIP application bandwidth exploitation can be improved by packet payload compression. In 2010, the Delta-Multiplexing method has been proposed to combine packet coalesces and packet payload compaction. Similar to the previous method, the Delta-Multiplexing works in environments with several VoIP LAN that are connected to the same VoIP WAN gateway. At the sender side VoIP WAN gateway, the received packet header and payload are separated from each other. Then, the payload is compressed to a smaller payload (s-pld), which gained by finding the delta between the consecutive packet's payload based on a certain mechanism and mathematical equations. After that, an RTP header along with a small header is added to each s-pld, which produces a small-packet (s-pkt). The purpose of the small header is to inspect the s-pkt within the coalesced packet at the receiver VoIP WAN gateway. Finally, the s-pkts that are traveling throughout the same path to the same destination VoIP WAN gateway are coalesced in one UDP/IP header and transmitted to their receiver. At the sender side VoIP WAN gateway, the opposite operations occur to restore the original VoIP packets. In which, the received coalesced packet is segregated to s-pkt, the payload restored to its original size, the original RTP/UDP/IP is attached to each payload, and the packets are sent to their final destinations. The performance evaluation of the Delta-Multiplexing showed that bandwidth exploitation has enhanced by up to 72% in comparison to the traditional method without coalescing the packets [3]. Another method that combines between packets coalesces and packet payload compaction has been proposed in 2019. The proposed method performs packet payload compaction with a similar way of the Delta-Multiplexing methods but over ITTP protocol. As for packet coalesce, it also works in same environments as Delta-Multiplexing and with similar general operations. However, only 8-bits small header is added to the compacted payload instead of 12-bits in Delta-Multiplexing. In addition, the s-pkts are coalesced at the network layer in an IP header instead of in the transport layer. The performance evaluation of the proposed method showed that bandwidth exploitation has highly enhanced in comparison to the traditional method of ITTP protocol [13].

As we can see, the pre-discussed research methods are focusing on improving the VoIP applications bandwidth utilization. This research paper proposes a new method, called Payload Shrinking and Packets Coalesce (PS-PC), which improves the VoIP applications bandwidth utilization as well. However, the proposed PS-PC method combines two techniques to improve VoIP applications bandwidth utilization. First, the proposed PS-PC method uses a new unique technique to coalesce a group of VoIP packets that are travelling to the same destination in one header, instead of a separate header to each packet. Second, the proposed PS-PC method uses a new unique technique that shrinks the VoIP packet payload to a smaller one by removing some bits from the start or end of the payload based on a certain algorithm. The combination of packets coalesces technique and packet payload shrinking technique has reduced the header overhead and reduce the size of the VoIP packet payload, respectively. Thus, highly improve VoIP applications bandwidth utilization. The following section discusses the proposed PS-PC method in details.

3. Payload shrinking and packets coalesce (PS-PC) method

This section discusses the suggested PS-PC method. The preferable location to deploy the PS-PC is a VoIP gateway that represents an exit point to a myriad number of simultaneous VoIP calls, which are traveling to the same destination VoIP gateway. Fig. 2 illustrates the PS-PC method location. As we can see, the PS-PC method has two main components, namely, Sender side PS-PC (S-PS-PC) and Receiver side PS-PC (R-PS-PC) components. Section 3.1 discusses the internal process of S-PS-PC component. Section 3.2 discusses the internal process of R-PS-PC component.

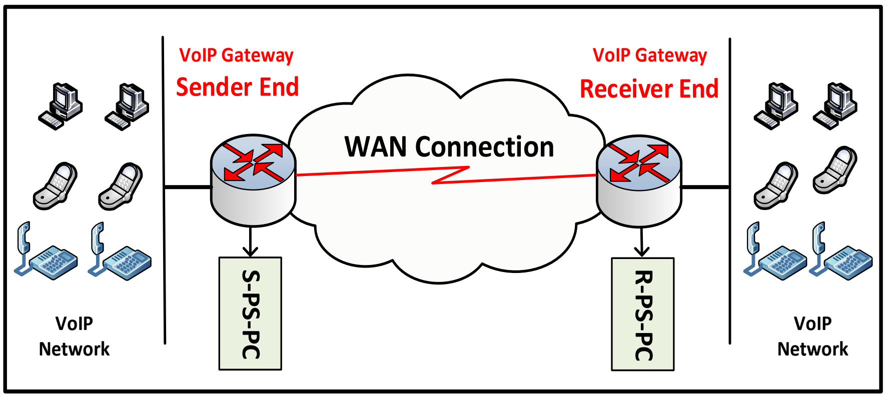

Fig. 2. PS-PC method components (S-PS-PC and R-PS-PC)

3.1. S-PS-PC component

Two main functions occur at the S-PS-PC component to ameliorate VoIP bandwidth utilization, namely, packets coalesce and packet payload compaction. The S-PS-PC component internal process goes through several steps to perform these two functions.

Step 1. Group the incoming VoIP packets based on their destination's VoIP gateway.

Step 2. Separate the VoIP packets header and payload.

Step 3. Shrink the payload to smaller one in size, called a runt-payload. The algorithm to perform the shrinking process and produce the runt-payload is discussed below.

Step 4. Specifies a unique ID to each call. This ID will be added to the header of each packet that belongs to this call. Therefore, this ID must be saved in a state table in both sender and receiver VoIP gateways.

Step 5. Attach the original ITTP header along with a new header, called runt-header, to each runt-payload. This will produce a small packet with a new format, called runt-packet. Fig. 3 shows the runt-packet format. The runt-header fields and their purpose will be discussed below.

Step 6. Concatenate the runt-packets of the same group together in one large VoIP packet payload, called giant-payload.

Step 7. Attach an IP header to the giant-payload which produces a large packet, called giant-packet. The source address of the IP header is the address of the sender VoIP gateway and the destination address of the IP header is the address of the receiver VoIP gateway. The giant-packet is sent to the receiver VoIP gateway. Fig. 4 shows the giant-packet format. Fig. 5 shows the internal process of the S-PS-PC component at the sender side VoIP gateway.

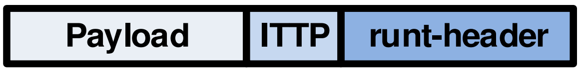

Fig. 3. Runt-packet format

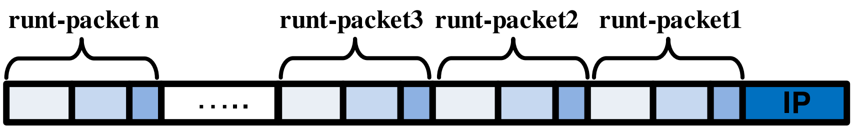

Fig. 4. Giant-packet format

As stated in Step 3, the VoIP packet payload is shrunk to a smaller payload (runt-payload). The shrinking algorithm based on the fact that the payload size is not changed during the entire call time. Thus, removing some bits from the start or end of the payload can be easily restored if designing a suitable algorithm. Fig. 6 depicts the shrinking algorithm at the S-PS-PC Component.

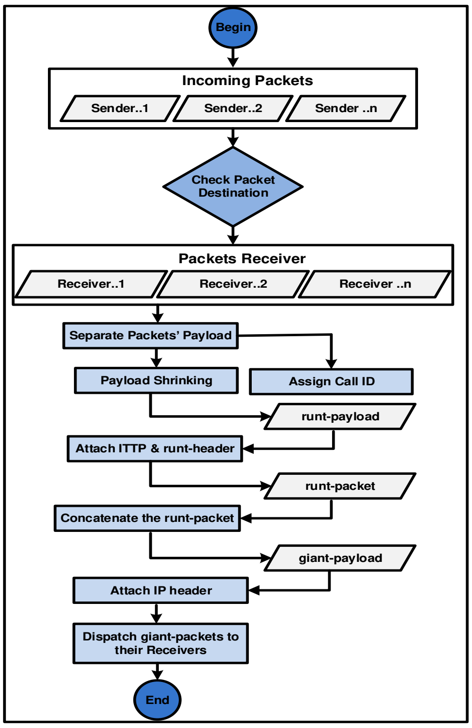

Fig. 5. S-PS-PC component internal process

Shrinking Algorithm@S-PS-PC Component

```
//pld is the payload of the incoming packet
//n is the number of zeros or ones at the start of pld
//m is the number of zeros or ones at the end of pld
//F1 and F2 are Boolean values
//SB is the shrunk bits field in the runt-header
//r-pld is the runt-payload

if (leftmost bit of pld = 0) {
    n= number of zeros at the start of pld
    F1 = 0 }
else if (leftmost bit of pld = 1) {
    n= number of ones at the start of pld
    F1 = 1 }
if (rightmost bit of pld = 0) {
    m= number of zeros at the end of pld
    F2 = 0 }
else if (rightmost bit of pld = 1) {
    m= number of ones at the end of P-pld
    F2 = 1 }
//---------------
if (n > m and F1 = 0) {
    r-pld = pld after eradicates leading zeros
    SB = 00 }
else if (n > m and F1 = 1) {
    r-pld= pld after eradicates leading ones
    SB = 01 }
else if (m >= n and F2 = 0) {
    r-pld = pld after eradicates trailing zeros
    SB = 10 }
else if (m >= n and F2 = 1) {
    r-pld = pld after eradicates trailing ones
    SB = 11 }
```

Fig. 6. Shrinking algorithm at the S-PS-PC component

3.2. R-PS-PC component

The main function of the R-PS-PC component is to construct the original VoIP packets and send them to their destinations. The internal process of the R-PS-PC component goes through several steps to perform this function.

Step 1. Separate the giant-packet to IP header and giant-payload.

Step 2. Inspect the runt-headers within the giant-payload to separate the runt-packets.

Step 3. Strip the runt-header and ITTP header of each runt-packet to obtain the runt- payload.

Step 4. Restore the runt-payload to its original size based on the information in the runt-header. The algorithm to produce the original payload will be discussed below.

Step 5. Restore the original addresses of the IP header based on the state table.

Step 6. Attach the ITTP header along with the IP header to the payload from the previous step, which reconstructs the original packet that has arrived at the sender side.

Step 7. Transmit the packets generated in step 6 to their destinations.

Fig. 7 shows the internal process of the R-PS-PC component at the receiver side VoIP gateway.

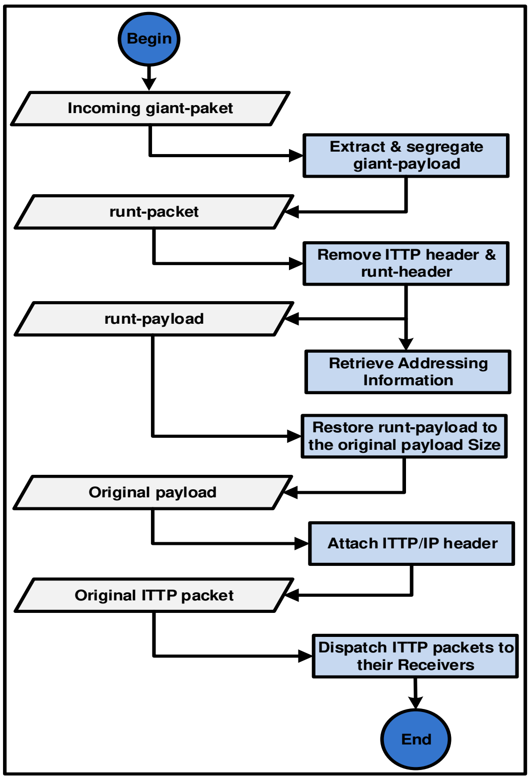

Fig. 7. R-PS-PC component internal process

As stated in Step 4, the shrinking algorithm restores the runt-payload to its original size, the size before applying the shrinking algorithm at the sender side VoIP gateway by the R-PS-PC component. Fig. 8 depicts the shrinking algorithm at the R-PS-PC Component.

Shrinking Algorithm@R-PS-PC Component

```
//pld  is the original payload
//r-pld is the runt-payload
//x is number of eradicated bits from pld
//L-pld is the length of the pld
//L-r-pld is the length of r-pld
//SB is the shrunk bits field in the runt-header

x = L-pld - L-r-pl
if (SB = 00)
    pld  = r-pld after adding x zeros at the start of r-pld
else if (SB = 01)
    pld  = r-pld after adding x ones at the start of r-pld
else if (SB = 10)
    pld  = r-pld after adding x zeros at the end of r-pld
else if (SB = 11) {
    pld  = r-pld after adding x ones at the end of r-pld
```

Fig. 8. Shrinking algorithm at the R-PS-PC component

3.3. Runt-header

As stated in Section 3.1 Step 5, a one-byte runt-header is attached to each runt-payload. The runt-header contains the necessary information to separate the runt-packet within the giant-payload and to restore the runt-payload to its original size (Step 2 and Step 4 in Section 3.2). Accordingly, the runt-header contains two fields, 6-bits ID and 2-bits Shrunk Bits (SB). Fig. 9 shows the runt-header format. The ID is selected for each call during the call setup. This ID will be added to the header of each packet that belongs to this call and it is used by the R-PS-PC component at the sender side VoIP gateway to find the address of the destination from the state table. The state table maps the destination address to its corresponding ID and is located at both S-PS-PC component at the sender side VoIP gateway and R-PS-PC component at the receiver side VoIP gateway. Table 1 demonstrates the ID and the corresponding address information within the state table. The ID of each runt-packet must be unique within the same giant-packet. Therefore, a giant-packet can have up to 64 mini-packets.

Fig. 9. Runt-header format

Table 1. State table

S-PS-PC component			S-PS-PC component	
ID	IP address: Port		ID	IP address: Port
8	12.10.0.1:20		8	12.10.0.1:20
11	12.10.0.2:21		11	12.10.0.2:21
12	12.10.0.3:27		12	12.10.0.3:27
13	12.10.0.4:42		13	12.10.0.4:42
16	12.10.0.5:19		16	12.10.0.5:19

The shrinking algorithm uses the SB field to denote the value (0 or 1) and place (start or end) of the removed bits from the VoIP packet payload. The 2-bits SB field have four different probability values. The value 00 means that the shrunk bits are zeros at the start of the packet payload, the value 01 means that the shrunk bits are ones at the start of the packet payload, the value 10 means that the shrunk bits are zeros at the end of the packet payload, and the value 11 means that the shrunk bits are ones at the end of the packet payload. Table 2 shows several examples that demonstrate the shrinking algorithm and how it uses the 2-bits SB field to denote the value and location of the shrunk bits in the VoIP packet payload. The meaning of n and m in the table is elaborated in Fig. 6. Notably, the values of the voice frame in Table 2 are not real and are only for demonstration purposes.

Table 2. Example of shrinking the payload

Original payload	n	m	Action	Runt-payload
0000001101100110010000	6	4	Shrink the leading zeros because n is greater than m	1101100110010000
0001110100100001000000	3	6	Shrink the trailing zeros because m is greater than n	0001110100100001
0000000110100011001111	7	4	Shrink the leading zeros because n is greater than m	110100011001111
1111111001110110010000	7	4	Shrink the leading ones because n is greater than m	001110110010000
1111101011001111111111	5	10	Shrink the trailing ones because m is greater than n	111110101100

4. Packets coalesce mathematical model

As mentioned earlier, the proposed PS-PC method combines two techniques to improve VoIP applications bandwidth utilization, namely packets coalesce technique and packet payload shrinking technique. In this section, we will provide mathematical equations that can be used to calculate the bandwidth utilization ratio when using the packets coalesce technique. However, packet payload shrinking cannot be calculated using mathematical equations because the pattern of the bits varies from a payload to another and, thus, the number of shrunk bits varies from payload to another. The bandwidth utilization ratio of the packets coalesce technique is calculated using the next equation:

(1) $$\mathrm{BW}_u = \frac{\mathrm{Pld}_S}{\mathrm{Pkt}_S} \times 100\%,$$

where, BW_U is the bandwidth utilization, Pld_S is the packet payload size and Pkt_S is the packet size in bytes. From Equation (1), we can derive the PS-PC method and the ITTP protocol bandwidth utilization ratio. The ITTP protocol bandwidth utilization ratio is calculated using

(2) $$\text{ITTP BW}_u = \frac{n\times\mathrm{Pld}_S}{n\times(\mathrm{Pld}_S + I_h + \mathrm{IP}_h)} \times 100\%,$$

where, n is the number of VoIP packets, I_h is the ITTP protocol header size and IP_h is the IP protocol header size in bytes. The PS-PC method bandwidth utilization ratio is calculated using

(3) $$\text{PS-PC BW}_u = \frac{\sum_{i=1}^{n}\mathrm{Pld}_S}{IP_h + \sum_{i=1}^{n}\mathrm{Pld}_S(i) + I_h(i) + R_h(i)} \times 100\%,$$

where, R_h is the runt header size in bytes. For example, assuming the number of VoIP packets is 10 and the VoIP packet payload size is 10 bytes, then the bandwidth utilization ratio of the ITTP protocol and PS-PC method are 27.7% and 50%, respectively.

5. PS-PC method performance evaluation

In this section, we will investigate the performance of the proposed PS-PC method, in terms of bandwidth usage compared to the traditional ITTP protocol (without packets coalesce or payload shrinking). We have used two elements to investigate the bandwidth usage, header overhead and consumed bandwidth. The header overhead of the two methods (PS-PC and ITTP) is displayed in Fig. 10. The result shows that the header overhead occurred when using the PS-PC method is less than the header overhead when using traditional ITTP protocol. This is because the proposed PS-PC method coalesces several packets in one IP header, unlike the traditional ITTP protocol, which attaches a separate header to each packet.

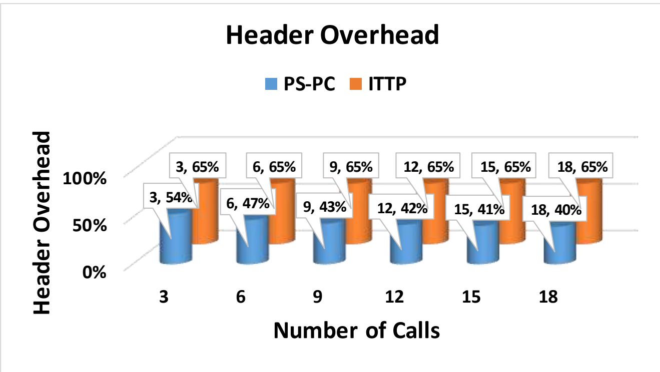

Fig. 10. Header overhead

The consumed bandwidth of PS-PC and ITTP methods is displayed in Fig. 11. The result shows that the bandwidth consumption occurring when using the PS-PC method is less than the bandwidth consumption occurring when using traditional

HTTP protocol. This is because the proposed PS-PC method coalesces several packets in one IP header and shrinks the VoIP packet payload, unlike the traditional ITTP protocol which does not apply any of them. According to the measured elements (header overhead and consumed bandwidth), the proposed PS-PC method presents better bandwidth usage than traditional ITTP protocol.

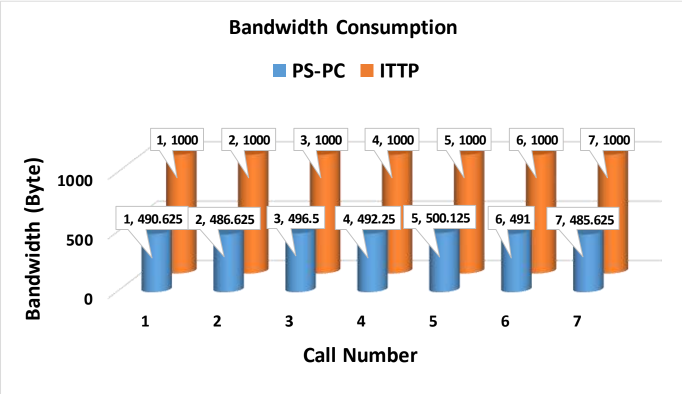

Fig. 11. Consumed bandwidth

6. Conclusion

VoIP is one of the most propagated technologies in the last few years. In this paper, we have investigated one of the key problems that the VoIP faces, which is the bad bandwidth utilization. We have proposed a new method to improve bandwidth utilization named PS-PC method. The PS-PC method works on two dimensions to improve bandwidth utilization: i) reducing header overhead through coalescing several VoIP packets in one IP header instead of a separate header to each packet and ii) shrinking the VoIP packet payload size to a smaller payload based on certain algorithm. The suggested PS-PC method has been deployed over the ITTP protocol. A runt-header has been added to the protocol structure in order to be able to perform both packets coalesce and VoIP packet payload shrinking. The bandwidth usage of the proposed PS-PC method was investigated in comparison with the traditional ITTP protocol without packets coalesce or payload shrinking. The result shows that the proposed PS-PC method presents better bandwidth usage than traditional ITTP protocol. As future work, the PS-PC method will be investigated in comparison to other coalesce methods. Moreover, other evaluation factors will be used to investigate the performance of the PS-PC method.

References

1. Gupta, A., A. Chaudhary. A Metaheuristic Method to Hide MP3 Sound in IPEG Image. – Neural Computing and Applications, Vol. **30**, 2018, No 5, pp. 1611-1618.
2. Zhang, Y., K. Xie, O. Ruan. An Improved and Efficient Mutual Authentication Scheme for Session Initiation Protocol. – PloS One, Vol. **14**, 2019, No 3, pp. e0213688.

3. A b u-A l h a j, M., M. K o l h a r, L. C h a n d r a, O. A b o u a b d a l l a, A. M a n a s r a h. Delta-Multiplexing: A Novel Technique to Improve VoIP Bandwidth Utilization between VoIP Gateways. – In: Proc. of 10th IEEE International Conference on Computer and Information Technology, 2010.
4. L i, X., H. H u. QoS Routing Algorithm Based on Entropy Granularity in the Network Transmission. – Cybernetics and Information Technologies, Vol. **19**, 2019, No 4, pp. 61-72.
5. A b u a l h a j, M., M. A l-T a h r a w i, S. A l-K h a t i b. A New Method to Improve Voice over IP (VoIP) Bandwidth Utilization over Internet Telephony Transport Protocol (ITTP). – In: Proc. of 8th International Conference on Software and Information Engineering, 2019, pp. 192-195.
6. S a n g w a n, A., M. C h i r a n t h, H. J a m a d a g n i, R. S a h, R. V e n k a e s h a, G. V i s h a l. VAD Techniques for Real-Time Speech Transmission on the Internet. – In: Proc. of 5th IEEE International Conference on High Speed Networks and Multimedia Communication (Cat. No 02EX612), 2002, pp. 46-50.
7. S a n d l u n d, K., G. P e l l e t i e r, J. L e. The RObust Header Compression (ROHC) Framework. RFC, 5795, 2010.
8. V a l i n, J., T. T e r r i b e r r y, C. M o n t g o m e r y, G. M a x w e l l. A High-Quality Speech and Audio Codec with Less than 10-ms Delay. – IEEE Transactions on Audio Speech, and Language Processing, Vol. **74**, 2015, No 1, pp. 58-67.
9. A t a n a s o v, I. Study on Deployment of Web Services for User Interaction in Multimedia Networks. – Cybernetics and Information Technologies, Vol. **13**, 2013, No 2, pp. 63-74.
10. S t e l i o s, A. S. Introducing VoIP Signaling. – Voip Signaling Protocols, February 2020. **https://www.pluralsight.com/blog/it-ops/**
11. K a u r, G., J. K a u r, S. A g g a r w a l, C. S i n g l a, N. M a h a j a n, S. K a u s h a l, A. K. S a n g a i a h. An Optimized Hardware Calibration Technique for Transmission of Real-Time Applications in VoIP Network. – Multimedia Tools and Applications, Vol. **78**, 2019, No 5, pp. 5537-5570.
12. S p e n c e r, M., et al. Iax: Inter-Asterisk Exchange. Version 2. – Internet Request for Comments, 2010.
13. A b u a l h a j, M. CA-ITTP: An Efficient Method to Aggregate VOIP Packets over ITTP Protocol. – International Journal of Innovative Computing, Information and Control (IJICIC'19), Vol. **15**, 2019, No 3, pp. 1067-1077.
14. O r t e g a, M., G. A l t a m i r a n o, C. B a r r o s, M. A b a d. Comparison between the Real and Theoretical Values of the Technical Parameters of the VoIP Codecs. – In: Proc. of IEEE Colombian Conference on Communications and Computing (COLCOM), 2019, pp. 1-6.
15. A b u a l h a j, M. ITTP-MUX: An Efficient Multiplexing Mechanism to Improve VOIP Applications Bandwidth Utilization. – International Journal of Innovative Computing, Information and Control (IJICIC), Vol. **11**, 2015, No 6, pp. 2063-2073.
16. V u l k a n, C., A. R a k o s, Z. V i n c z e, A. D r o z d y. Reducing Overhead on Voice Traffic. U.S. Patent 8,824,304, Issued 2 September 2014.

Received: 27.02.2020; Second Version: 02.06.2020; Accepted: 25.06.2020