Tel-MPLS: a new method for maximizing the utilization of IP telephony over MPLS networks

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Article Info

ABSTRACT

Currently, the multiprotocol label switching (MPLS) standard is extremely prevalent. By exploiting the features provided by MPLS technology, a range of services, including IP telephony, have enhanced their overall performance. However, due to the size of the packet header, the IP telephony service consumes a significant portion of the MPLS network's available bandwidth. For instance, in IP telephony over MPLS networks, the packet header might account for as much as 80% of lost time and bandwidth. Designers working on IP telephony are making substantial efforts to address this issue. This study contributes to current efforts by proposing a novel approach called Tel-MPLS, which involves IP telephony over MPLS. Tel-MPLS approach uses the superfluous fields in the IP telephony packet's header to retain the packet data, therefore lowering or zeroing the IP telephony packet's payload. Tel-MPLS is an approach that significantly reduces the bandwidth of IP telephony MPLS networks. According to the findings, the Tel-MPLS approach is capable of reducing the amount of bandwidth that is lost by 12% when using the G.729 codec.

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1. INTRODUCTION

The term wide area network (WAN) refers to any network that spans across multiple geographic locations. Existing WANs are built for a variety of applications that affect practically every aspect of contemporary life. Several WAN protocols, including point-to-point, multiprotocol label switching (MPLS), and frame relay, have been developed over time [1], [2]. MPLS avoids time-consuming routing database lookups that require lengthy network identifiers by using short path labels and thus optimizes the routing process. MPLS delivers a number of characteristics that improve a vast array of services and technology. These features include enhancements to network uptime, bandwidth usage, and network congestion reduction. IP telephony is a service that utilizes MPLS and its features [3]–[5]. IP telephony is a technology that enables phone and video calls to be made and received via the internet instead of landlines. With an internet connection, it is possible to make phone calls without the need for traditional phone service or copper lines. To make calls, one needs solely a high-speed internet connection and an IP telephony service provider [6]–[8]. Due to the aforementioned characteristics, MPLS technology has helped promote IP telephony [9]–[11].

Despite these potential benefits, IP telephony is now wasting a substantial percentage of the bandwidth made accessible by MPLS. This problem arises because the IP telephony packet's header is relatively long compared to the content [11], [12]. One example is the G.729 codec, which wastes up to 80% of the authorized bandwidth for an IP telephony call, as shown in Figure 1 [13]–[15]. It is evident that all codecs result in a considerable amount of useless bandwidth. The issue of MPLS network resource waste must be taken into consideration and properly investigated. The creators of IP telephony have devised a number of approaches for addressing IP telephony's big header. Examples of these approaches include putting the IP telephony packet into a one header, compressing the packet's header, substituting the protocols in the IP telephony header with new protocols pertaining to IP telephony, and utilizing the extra fields in the packet's header to convey the voice data [14]–[18]. A key benefit of the last approach, which uses the superfluous header fields, is the existence of a number of recommended solutions that are consistent with the existing standards and equipment [18]–[20]. However, none of these methods have been optimized for IP telephony over MPLS networks, as proposed in this study.
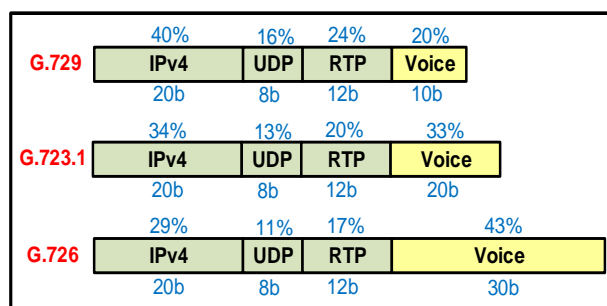

Figure 1. IP telephony packet

2. RELATED WORK

IP telephony's propagation has been hindered for some time now due to bandwidth constraints. There have been several suggested solutions to circumvent this issue and quicken the migration to IP telephony. One of the first proposed solutions is the packet/frame aggregation standard, which entails encapsulating all packets that travel along the same route into a single header. Within the constraints of this packet/frame aggregation standard, a vast array of alternative approaches have been created. Some of these approaches were designed to aggregate packets at layer 3 in one header, while others were designed to aggregate frames at layer 2 in one header. Nonetheless, these techniques all belong to the category of "frame aggregation." Under the category of packet aggregation, a considerable variety of methods have been established [21]–[24]. As an example, Seytnazarov and Kim [21] proposed a multiplexing approach for IEEE 802.11n wireless networks. IP telephony frames are multiplexed into a single aggregation media access control (MAC) protocol data unit (A-MPDU) at layer 2 of the open system interconnection (OSI) model. If a single frame inside the A-MPDU is corrupted, only that frame is retransmitted. Furthermore, the approach adjusts the size of the multiplexed frames based on channel load, real-time transport protocol/real-time control protocol (RTP/RTCP) reported delay, buffering delay, 150 ms delay, and average access latency to the medium. The simulation showed that the approach outperforms comparable methods. However, the multiplexing group has several drawbacks. Distinct packets from multiple sessions are multiplexed into a single chunk to receive the intended service. In reality, certain sessions should provide superior service compared to others. Second, IP telephony apps employ certain methods to mask missing packets and enhance call clarity. Small IP telephony packets are effective with the hiding methods. Nonetheless, the hiding methods will not be successful with a substantial portion of multiple packets. As a consequence, the missing packet is not disguised and call clarity suffers. Third, increasing the number of packets that are multiplexed into a single chunk will enhance the bandwidth's efficiency. If the number of sessions is limited, the packets must remain in the buffer waiting, for a sufficient number of packets to arrive in order to permit multiplexing. In addition, the duration of the multiplexing process depends on the approach employed. Due to the delay induced by the buffer's waiting time and the multiplexing process, the call's clarity will be compromised. Lastly, the multiplexing activity depletes the resources of the multiplexing device [22], [25], [26].

A method known as header compression can be applied to minimize the length of the packet's header. This method reduces the size of the header by deleting individual fields. Because they adhere to predefined patterns, the receiving end of the call can readily deduce their values, allowing them to gather their data. When IPv4 is employed, the header is shortened to as small as 2 bytes, therefore conserving a

substantial amount of bandwidth. Fortuna and Ricardo [27] have developed a model to examine the performance of robust header compression (RoHC) when operating IP telephony across 802.11 channels. The design model selected the RoHC U-mode. An additional component, RoHC Gain, was suggested to measure the bandwidth increase that other streams can consume as a result of RoHC employment with IP telephony traffic. Research shows that RoHC should only be used when 802.11 connections are crowded or transmitting greedy flows [27], [28]. Regardless of the technique employed, the header compression category has a number of problems. First, header compression approaches perform poorly in environments with substantial packet loss or lengthy round-trip times. Second, header compression requires several processes that were taxing compression/decompression devices to the point where their resources were being squandered. In addition, the packet compression/decompression procedures will introduce a new source of latency for IP telephony applications [29], [30].

Several IP telephony software developers have proposed the development of new IP telephony-specific protocols, including the inter-asterisk exchange (IAX) protocol. IAX is an underlying communications protocol that is incorporated into the asterisk private branch exchange (PBX) software. IAX is also supported by a variety of other soft-switches, PBX systems, and softphones. It is utilized to transport IP telephony traffic between servers and endpoints. IAX is an IP telephony protocol that can handle any kind of streaming media, including video, but focuses primarily on IP voice calls. The IAX protocol employs a single protocol for simultaneously managing and sending media. In addition, its open architecture allows for the incorporation of new payload kinds, which is necessary to provide more services. IAX substitutes its own 4-byte mini-header for RTP's 12-byte protocol in order to transmit digital speech data. In addition, it employs minimal encoding that decreases bandwidth use, making it a good alternative for internet telephone services [14], [31].

Utilizing the header's redundant fields to transmit the packet's voice data is one of the recently examined potential options. The notion that various types of application data are exchanged via the IP telephony packet header protocol forms the basis of the proposed method. As a result, particular apps require particular fields contained within certain protocols, while other apps do not require them. The latter apps put needless pressure on the available bandwidth and served no purpose. These fields have been designated as superfluous. This last solution to the bandwidth issue utilizes the superfluous fields within the packet to transport the voice data. Consequently, the payload can be lowered or discarded altogether, resulting in bandwidth savings. The short voice frame (SVF) technique suggested in [32] is one of the most efficient ways to execute this solution. Based on a number of key assumptions, the SVF method is able to efficiently use the seven unused packet header fields. An IPv4 telephony packet may be 17 bytes shorter than the standard 50-70-byte size. The results of the research show that the amount of bandwidth saved can reach 29% in some cases [32].

However, bandwidth problem impacting IP telephony and the IP protocol remain unresolved, particularly in MPLS networks. This research project devises a solution to this problem by utilizing the superfluous fields present within the IP telephony packet header. IP telephony over MPLS (Tel-MPLS) is the method meant to decrease the payload size of IP telephony packets to the greatest extent feasible.

3. METHOD

The operating mechanism of the Tel-MPLS technology is dissected and explored in this section. When IP telephony is utilized, the fundamental purpose of the Tel-MPLS strategy is to maximize the channel capacity (Ch-C) of the MPLS network. Tel-MPLS is a method utilized by calling clients (such as Zoiper) and intermediary devices. Using the Tel-MPLS method, the Ch-C of either one may be raised with the same degree of efficiency. It has been assumed for the purposes of this study that the Tel-MPLS technique is applicable to the calling client. Tel-MPLS may be implemented using the topology represented in Figure 2.

3.1. Core idea

Tel-MPLS increases call capacity by using IP telephony packet header fields that are unused. Superfluous fields are employed in order to save digital voice data within the packet. Several studies used certain rules to identify redundant fields in the IPv4 telephony packet for IPv4 telephony applications [6], [18], [19], [32]. Tel-MPLS, on the other hand, is designed for use with IP telephony in all circumstances (no specific rules). As a result, only the source IP address and the source port number (source socket) were deemed superfluous information and were utilized to keep the voice payload of the packet, as will be described in the following sections.

3.2. Source socket

Source sockets identify data senders in packet-based networks. Receivers use this source socket to reply to senders [19]. The initiation of an IP telephony call involves two steps: signaling and media

transmission. During the signaling phase, a specific protocol, such as the H323 protocol, is used to initiate the call and agree on its specifications, including the sockets of each call end. At this point, it is possible to say that each call is aware of the socket on the other end. During the media transmission phase, a separate protocol, such as RTP, is used to send voice data [8], [33]. RTP, along with user datagram protocol (UDP) and IP, transfers voice data to its intended destination, utilizing the call factors from the initial phase, including the source socket. The key point is that every single voice data packet contains the source socket. IP telephony media transmission sessions, on the other hand, are not request-and-response exchanges, and the callee client does not need to respond to the sender. If the callee responds, the callee client will utilize the source socket generated at session start [29], [32], [34]. Furthermore, this issue does not impact other network devices such as switches, routers, and firewalls. The switch operates on the second layer of the OSI model, whereas IP operates on the third. The IP address of the destination is utilized by the router for its main routing function, and when H323 IP telephony signaling protocols are in use, a firewall will only block an incoming call during the call setup procedure. Furthermore, this design will not have an impact on any of the other protocols. Therefore, the source socket address that is included in voice data packets is superfluous and may be reclaimed to serve another function, such as transmitting the voice data itself.

3.3. Carrying the voice data

The Tel-MPLS technique transmits voice data through the source socket. The sender's client first extracts the voice data from the packet. The source socket field is subsequently filled in with the voice data [34]. As a result, these variables appropriately indicate the size of the payload. Figure 3 displays the pseudocode for the mechanism the IP telephony client uses on the sender side. Lines 1 to 4 contain the identification of the variables that comprise the pseudocode. Line 6 inserts the voice data into the source socket fields. Altering the value of the payload length field in the IP header is achieved through the use of line 7. The value of the length field in the UDP header may be changed by the usage of line 8. The voice data is restored from the source socket by the IP telephony client, which then inserts it in the playout buffer located on the receiver's side. Without the padding, only the voice data itself must be recovered. In order to choose the duration of the voice data, the IP telephony client must determine the codec that is currently being used. During the signaling phase of an IP telephony call, the codec to be used is specified. Figures 4 and 5 show in detail the process performed by the IP telephone client on the sending and receiving sides of the call.
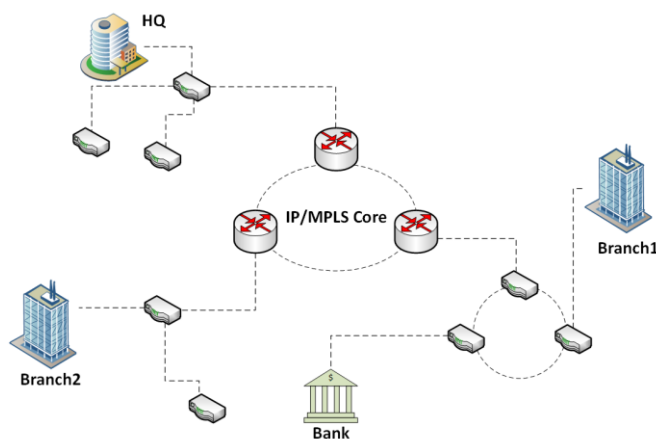

Figure 2. Tel-MPLS method topology

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Tel-MPLS Method→ Sender IP Telephony Client

1 //L is length of voice data in bits //Integer
2 //SrcSkt is the source socket address // String
3 //L-Pld is the value of Payload Length field in the IP header //Integer
4 //Len is the value of Length field in the UDP header //Integer

5 For each IP Telephony packet {
6        Place 6 bytes in the SrcSkt;
7        L-Pld= 20 + (L / 8 - 6)
8        Len= 12 + (L / 8 - 6)
9     }
```

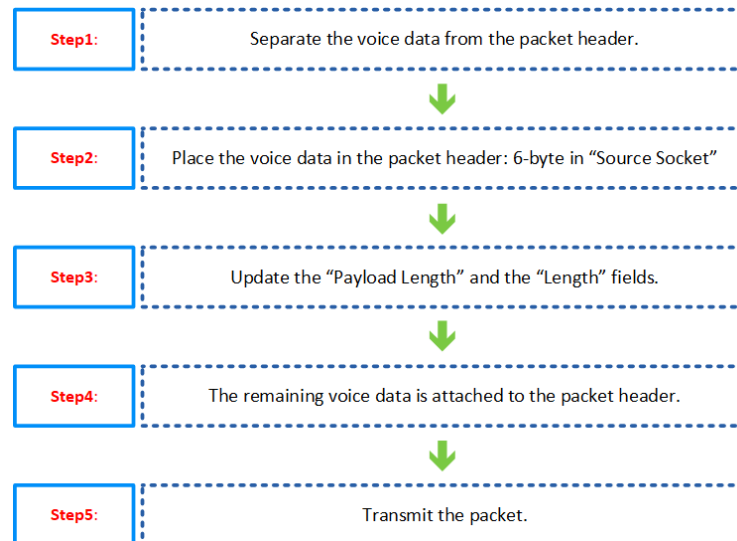Figure 3. Tel-MPLS technique algorithm-sender side

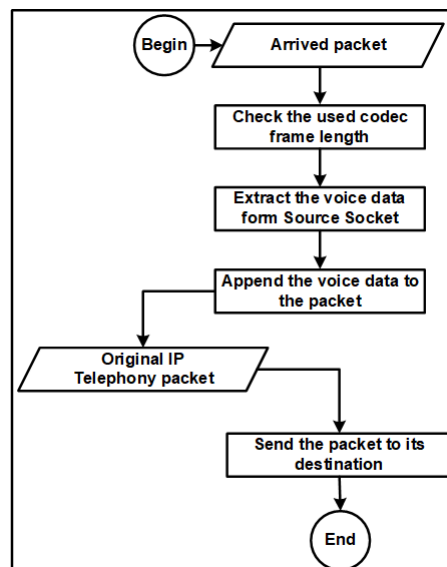Figure 4. Tel-MPLS technique process-sender side

Figure 5. Tel-MPLS technique process-receiver side

4. RESULTS AND DISCUSSION

The Tel-MPLS technique is compared to the original RTP (O-RTP) method, which is used to transport IP telephony packets. Comparisons are made between the proposed Tel-MPLS technology and the SVF and O-RTP techniques. The SVF approach is chosen as it is comparable to the planned Tel-MPLS method, since the SVF uses the packet header's fields that are superfluous to carry voice data.

4.1. Channel capacity

Each channel with a bandwidth between 100 and 1,000 kb has had its Ch-C analyzed. The fact that packets are being dropped shows that the channel is overloaded. In accordance with this fact, the Ch-C is equal to the number of calls that may made before a packet is dropped, and it is measured accordingly. In Figures 6-8, the Ch-C of the Tel-MPLS technique is compared to the SVF and O-RTP methods using the G.723.1, G.726, and G.729 codecs, respectively. The Ch-C attained with the Tel-MPLS technique leads to a larger number of successful calls when compared to the O-RTP method. As bandwidth availability increases, the gap between Ch-C and available bandwidth widens. This issue has emerged due to the deletion of the payload through the use of an unneeded field (source socket) in the header to communicate the payload. In contrast, in terms of Ch-C, the SVF approach outperforms the Tel-MPLS method.
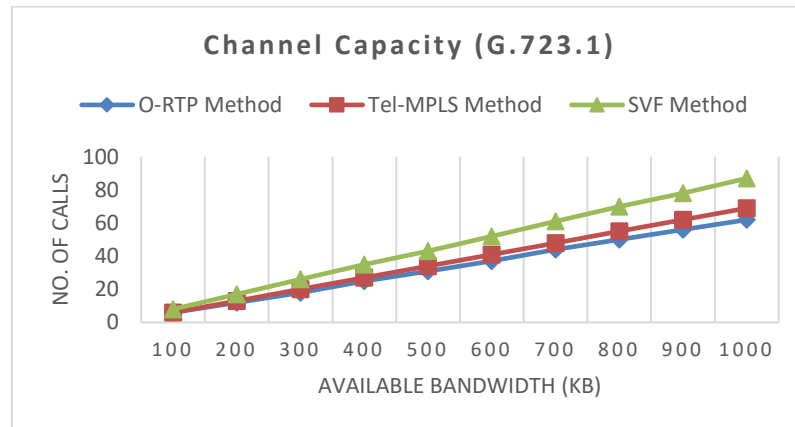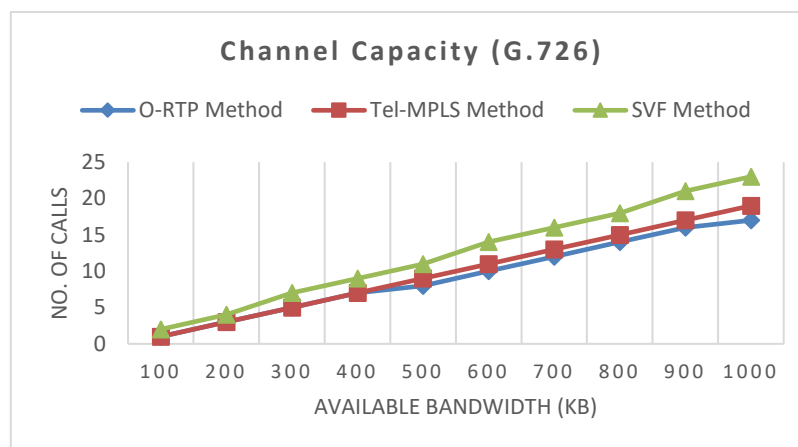
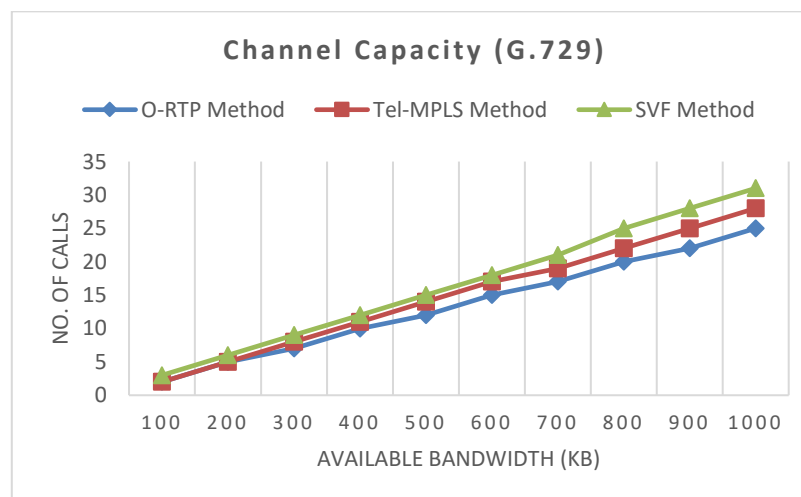Figure 6. Ch-C (G.723.1 codec)

Figure 7. Ch-C (G.726 codec)

Figure 8. Ch-C (G.729 codec)

4.2. Reduction in wasted bandwidth

In contrast to the SVF and O-RTP methods, the Tel-MPLS technique is analyzed with a number of calls between 10 and 100. As seen in Figure 9, the Tel-MPLS technique uses less total bandwidth than the O-RTP method. Compared to the O-RTP technique, employing the G.729 codec, the Tel-MPLS method

reduces the required bandwidth by approximately 12%. Alternatively, the SVF approach outperforms Tel-MPLS in terms of reduction in wasted bandwidth (R-BW).

Figure 9. Bandwidth reduction

In terms of Ch-C and R-BW, the SVF technique outperforms Tel-MPLS. However, the SVF approach employs several unnecessary fields to transport voice data. This limitation limits their applicability to the circumstances for which they were intended. The Tel-MPLS technique, on the other hand, employs only the source socket fields to convey the voice data. Because of this criterion, the Tel-MPLS approach becomes more broad and adaptable to any given situation. Compared to Tel-MPLS, the SVF approach puts more load on the network for various reasons. The SVF technique is intended to function on gateway routers. Therefore, these routers' resources are depleted, especially considering the large volume of calls. The Tel-MPLS approach is intended to operate on the client side without taxing the routers. The Tel-MPLS approach employs just the source socket to transmit voice data without resetting these fields to their original values. However, the SVF method uses multiple fields to convey the voice data, necessitating additional processes and calculations, such as resetting the superfluous field values at the receiver gateway. Performing these processes on every call packet uses the routers' resources, particularly when a large number of calls are in progress.

5. CONCLUSION

IP telephony and MPLS integration results in the loss of up to 80% of an MPLS network's potential capacity. With the Tel-MPLS strategy presented in this paper, we hope to contribute to the ongoing effort to find a solution to this issue. The sugested method effectively uses the source socket to transmit the packet's digital voice data, thereby reducing the IP telephony packet payload. Voice data is extracted from the packet and placed in the source socket fields by the sender client. As with any other payload, the IP telephony client at the receiver collects voice data from the source socket fields and appends it to the end of the packet. Using three unique codecs (G.729, G.726, and G.73.1), the O-RTP method is compared to the Tel-MPLS method. When employing the G.729 codec, the Tel-MPLS method reduces the amount of wasted bandwidth by 12% compared to the O-RTP technique.

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