A. Kayed, M.M. Abualhaj, M. Alharibat. Performance comparison of IAX and ITTP VoIP protocols. International Journal of Academic Research Part A; 2013; 5(2), 186-193. ***DOI:*** *10.7813/2075-4124.2013/5-2/A.29*

PERFORMANCE COMPARISON OF IAX AND ITTP VoIP PROTOCOLS

Ahmad Kayed[1], Mosleh M. Abualhaj[2], Mohammad Alharibat[3]

[1]Computer Science Department, Faculty of Information Technology, Middle East University (MEU), Amman,
[2]Computer Information System Department, Faculty of Information Technology,
Middle East University (MEU), Amman **(JORDAN)**
E-mails: akayed@meu.edu.jo, mabualhaj@meu.edu.jo, mharibat@meu.edu.jo

DOI: 10.7813/2075-4124.2013/5-2/A.29

ABSTRACT

VoIP is a technology used to transfer the voice calls over the Internet. There are many VoIP protocols proposed to transfer the voice calls over the Internet such as ITTP and IAX. This paper provides a comparative study for the ITTP and IAX protocols accompanied with experimental analysis and discussion. The experimental results showed that the ITTP outperformed the IAX protocol in terms of bandwidth utilization and voice quality (delay and packet loss). For example, when taking an 8 kbps codec with a 20 ms packetization and 20 byte packet size, the per-call bandwidth consumption of ITTP is 0.024% less then IAX and the Goodput is 11.7% more at 350kpbs bandwidth, which indicate better bandwidth utilization. On the other hand, the packet loss has reduced up to 10% and delay has reduced on average 15.1% in the tested cases, which indicts better voice quality.

Key words IAX, ITTP, and VoIP

1. INTRODUCTION

VoIP technology is becoming increasingly important as they gain market share in PSTN technology. In the short run, it is likely that VoIP technology will dominate the telecommunications market and replace the PSTN technology. Under the auspices of the Internet, VoIP provides many advantages that make it replace PSTN. One of these advantages is the cheap VoIP call rate. Another advantage would be the free services which can be integrated with VoIP such as voicemail three-way calling …etc. [1]. However, VoIP still suffer from some problems such as inefficient use of bandwidth, which will in turn degrade the VoIP quality [2]. The main source of inefficient bandwidth utilization is the VoIP transfer protocols such as RTP, IAX, and ITTP [Ref]. RTP is specialized to transfer all types of real-time media data, including voice, video, IPTV,…etc [3]. IAX, specifically the IAX mini-frame, can transfer real-time media data as well, and is highly optimized for VoIP calls [4]. Recently, the ITTP protocol emerged to transfer the VoIP calls as well [2]. However, both RTP and the IAX mini-frame are unable to transfer media data, including the VoIP calls, by themselves, which explains why they work on top of transport layer protocols. RTP and the IAX mini-frame typically work in conjunction with the transport layer UDP to transfer VoIP applications data. In contrast, ITTP is standalone protocol that can transfer the VoIP calls by itself [2] [5]. As RTP had been studied by many other researches, this work will provide a performance analysis of the ITTP in comparison with IAX with particular consideration to bandwidth utilization.

2. COMPARATIVE STUDY: IAX AND ITTP

In this section, we will provide a comparative study between the IAX and ITTP protocols. We will show in brief the main features of both protocols. In addition, we will show the amount of bandwidth consumed by the two protocols using mathematical equations, as this study focuses on the bandwidth utilization of each protocol and how it affect the VoIP quality.

2.1. IAX

IAX is a simple VoIP protocol that support features more than any other VoIP protocols. Unlike ITTP, which supports only voice media transfer, the IAX can supports both signalling and media transfer.in addition, IAX is a binary protocol designed to reduce overhead, especially with regards to voice streams. IAX's capabilities include traversing firewalls easily avoiding network address translation (NAT) traversal complications. As regards multiple calls, IAX reduces the overhead of each channel by combining data from several channels into one packet, thus reducing both the header overhead and number of packets at the same time [6].

However, IAX main purpose is to transfer the point-to-point VoIP calls, with the ability to handle most types of the media streams. The IAX includes many types of messages, called frame, for different purposes. The 4 bytes IAX mini-frame used to transfer the media data, figure 1 shows the IAX mini-frame header format. The IAX mini-frame was designed to be simple and reduces both overhead and bandwidth consumption resulting from 12 bytes RTP. Like RTP, The IAX mini-frame works in conjunction with the UDP protocol in order to transport the VoIP packets, figure 2 shows the IAX mini-frame and UDP combination. However, the packet overhead results from the 4 bytes IAX mini-frame and the 8 bytes UDP protocol between of 40% to 120% in the typical situations, which is substantial overhead. Most importantly, the IAX mini-frame is specified for IAX applications, and does not work with SIP or H323 which they are dominating the VoIP applications [4] [7]. Therefore, the chances to spread and use the IAX mini-frame by the VoIP applications are very limited.

IAX HEADER			
BITS	00	15	16 31
0	F	SOURCE CALL NUMBER	TIMESTAMP
32	DATA		

Fig. 1. *IAX mini-frame header format*

IAX/UDP HEADER			
BITS	00	15	16 31
0	F	SOURCE CALL NUMBER	TIMESTAMP
0	SOURCE PORT NUMBER		DESTINATION PORT
32	LENGTH		CHECKSUM
64	DATA		

Fig. 2. *IAX mini-frame and UDP headers combination*

2.2. ITTP

On the other hand, ITTP has a simple design that is highly optimized for transfer VoIP calls media only, figure 3 shows the ITTP header format. ITTP supplies VoIP applications with only the key information that is needed for functionality. All other functionalities are added on the layer above. In addition, ITTP combines the two functions, data transporter and voice media data carrier which performed by UDP and IAX mini-frame respectively, into one protocol in the transport layer [2].

The simplicity of the 6 bytes ITTP protocol addresses bandwidth utilization problem caused by the 12 bytes IAX/UDP protocols, which will in turn improve VoIP quality, as we will see in the implementation results section. Wherein, the packet overhead in the ITTP protocol is minimal, which maximized the bandwidth utilization. The state and processing overhead are also minimal, creating only a minimal delay and improving the quality of VoIP applications. The small ITTP header improves the buffer utilization, which reduces the packet loss ratio and improving the quality of VoIP applications as well.

In addition, this simple design gives opportunity to the VoIP developer to add suitable functions and methods for specific application requirements and purposes. On the other hand, ITTP will not be restricted to any signaling protocol, which gives it the opportunity to spread and be adopted by any signaling protocol [2].

ITTP HEADER			
BITS	00	15	16 31
0	F	SOURCE-PORT NUMBER	DESTINATION-PORT NUMBER
		TIMESTAMP	

Fig. 3. *ITTP header format*

2.3. ITTP and IAX/UDP Per-Call Bandwidth Consumption

The Per-call bandwidth consumption is the amount of bandwidth consumed when a VoIP call is made. The Per-call bandwidth consumption depends on the size of the frame and the size of the packet header. Equation 1 is used to calculate the per-call bandwidth consumption [8]:

$$\boldsymbol{C_{bw} = Pkt_s * PPS} \tag{1}$$

Where, C_{bw} is the per-call bandwidth in kbps, Pkt_s is the packet size in bytes, PPS is the number of packets per second. Equation is used to calculate the packet size:

$$\boldsymbol{Pkt_s = P_h + F} \tag{2}$$

Where, P_h is the packet header size and F is the frame size. Equation 5.9 is used to calculate the number of packets per second [8]:

$$\boldsymbol{PPS = \frac{C_{br}}{F}} \tag{3}$$

Where, C_{br} is the codec bit rate in second. From 5.7, 5.8, and 5.9 we can obtain the equations to calculate the call bandwidth consumption of the ITTP protocol and the RTP/UDP protocols. Equation 5.10 is used to calculate the ITTP protocol call bandwidth consumption:

$$\boldsymbol{ITTP_{C_{bw}} = Pkt_s * PPS}$$

$$\boldsymbol{= (P_h + F) * \frac{C_{br}}{F}}$$

$$\boldsymbol{= \frac{\left((I_h + IP_h + F) * \frac{C_{br}}{F*8}\right) * 8}{1000}} \qquad (4)$$

Where, I_h is the ITTP protocol header size and IP_h is the IP protocol header size. Equation 5.22 is used to calculate the IAX/UDP protocols call bandwidth consumption:

$$\boldsymbol{IAX/UDP_{C_{bw}} = Pkt_s * PPS}$$

$$\boldsymbol{= (P_h + F) * \frac{C_{br}}{F}}$$

$$\boldsymbol{= \frac{\left((X_h + U_h + IP_h + F) * \frac{C_{br}}{F*8}\right) * 8}{1000}} \qquad (5)$$

Where, X_h is the IAX protocol header size and U_h is the UDP protocol header size. Based on equations 5.10 and 5.22, Table 1 shows the calculation of the per-call bandwidth consumption of the ITTP protocol and IAX/UDP protocols using 8kbps codec bit rate, with 10 bytes, 20 bytes, and 30 bytes frames size.

Table 1. ITTP and IAX/UDP protocols per-call bandwidth consumption

Protocol	Frame Size (byte)	Per-call consumption bandwidth (kbps)	
ITTP	10	= (((10+6+20) * (8000/(10*8)))*8)/1000)	= 28.8
IAX/UDP		= (((10+4+8+20) * (8000/(10*8)))*8)/1000)	= 33.6
ITTP	20	= (((20+6+20) * (8000/(20*8)))*8)/1000)	= 18.4
IAX/UDP		= (((20+4+8+20) * (8000/(20*8)))*8)/1000)	= 20.8
ITTP	30	= (((30+6+20) * (8000/(30*8)))*8)/1000)	= 14.9
IAX/UDP		= (((30+4+8+20) * (8000/(30*8)))*8)/1000)	= 16.5

Figure 4 depicts the IAX/UDP protocols and the ITTP protocol per-call bandwidth consumption, using the result calculated in Table 1.

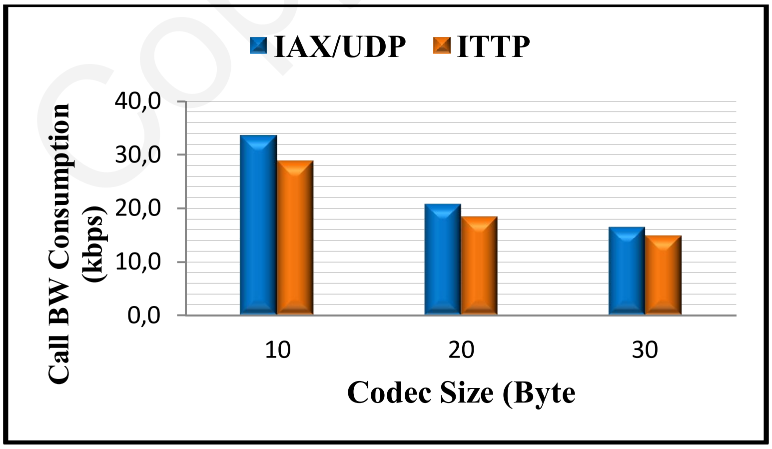

Fig. 4. *ITTP and IAX/UDP protocols per-call bandwidth consumption*

As we can see from Figure 4, in the case of 10 bytes frame size, the per-call bandwidth consumption of the IAX/UDP protocols is 33.6 kbps whilst the per-call bandwidth consumption of ITTP protocol is 28.8 kbps. In the case of 20 bytes frame size, the per-call bandwidth consumption of the IAX/UDP protocols is 20.8 kbps whilst the per-call bandwidth consumption of ITTP protocol is 18.4 kbps. In the case of 30 bytes frame size, the per-call bandwidth consumption of the IAX/UDP protocols is 16.5 kbps and the per-call bandwidth consumption of ITTP protocol is 14.9 kbps. As a result, with the three different frame sizes the ITTP protocol shows less per-call bandwidth consumption compared to the IAX/UDP protocols. This is due to the difference in the protocol header size.

3. IMPLEMENTATION ENVIRONMENT

The NS2 was used to simulate the ITTP protocol. NS2 is an open source simulation that continuously enhances and extends, upon control by the network developer and researcher, to support more and more network components. NS2 supports a vast number of network protocols, algorithms, applications, and so on. After simulating the ITTP, two different network topologies were designed using NS2 to demonstrate its performance. The G.729 codec with sample size 20bytes and bit-rate 8kbps was used in all experiments. The data rate upon using IAX/UDP and ITTP is 20.8 Kbps and 18.4 Kbps, respectively. The rate varies because of the different in the protocols header size. The specification of the two topologies as follow:

First topology:

Number of PCs 60
Number of routers 5
LAN links delay 2 milliseconds
WAN links delay 18 milliseconds
Queue DropTail with size of 50
Traffic generator CBR generator
Traffic starting time 100 milliseconds
Traffic ending time 1000 milliseconds

In the first topology, two experiments were executed to investigate the number of calls supported and Goodput of the ITTP protocol and compare it with the IAX/UDP protocol. Figure 5 depicts topology 1 and its basic configuration.

Second topology:

Number of PCs 10
Number of routers 6
LAN links delay 5 milliseconds
WAN links delay 20 milliseconds
Queue DropTail with size of 50
Traffic generator CBR generator
Traffic starting time 100 milliseconds
Traffic ending time 1000 milliseconds

In the second topology, two experiments were executed to investigate the Packet loss and Delay of the ITTP protocol and compare it with the IAX/UDP protocol. Figure 6 depicts topology 2 and its basic configuration. In experiment 1(Packet loss experiment), the PCs start and end transmitting the data as follow:

PC1 start time 0.05 seconds and end time 1.05
PC2 start time 0.16 seconds and end time 1.16
PC3 start time 0.275 seconds and end time 1.275
PC4 start time 0.385 seconds and end time 1.385
PC5 start time 0.491 seconds and end time 1.491

In experiment2 (Delay experiment), PCs start and end transmitting the data as follow:

PC1 start time 0.05 seconds and end time 1.1 seconds
PC2 start time 0.06 seconds and end time 1.16 seconds
PC3 start time 0.07 seconds and end time 1.22 seconds
PC4 start time 0.08 seconds and end time 1.28 seconds
PC5 start time 0.09 seconds and end time 1.34 seconds

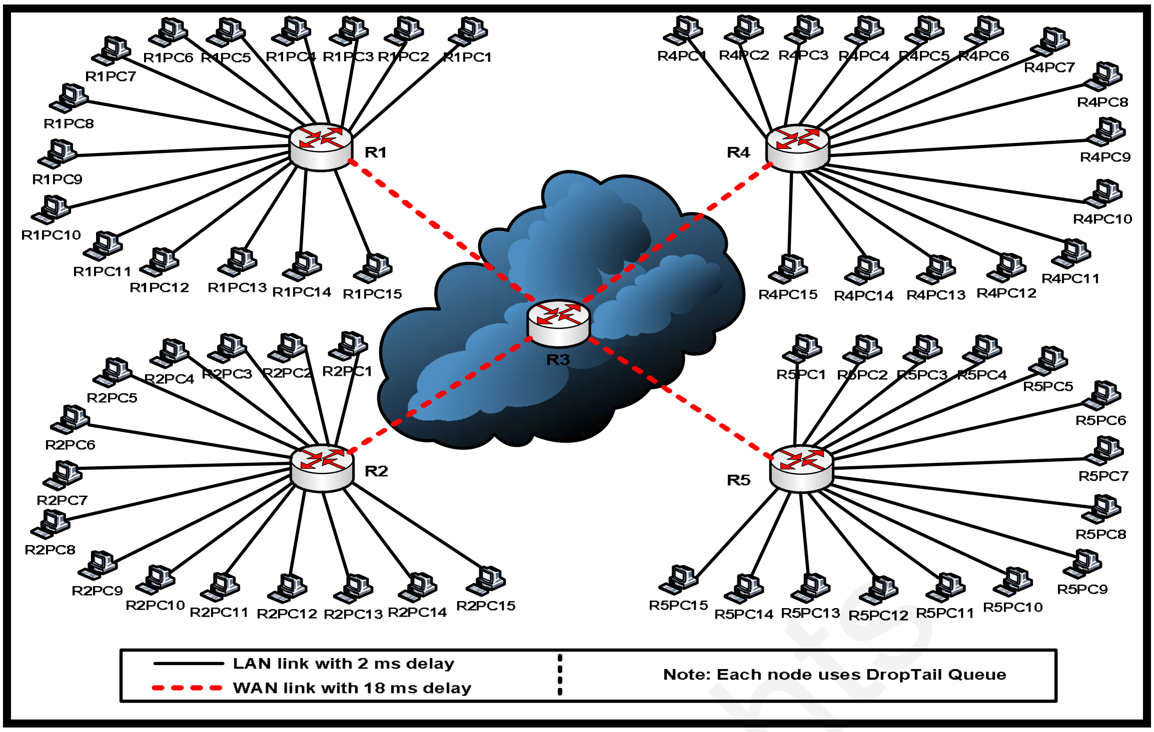

Fig. 2. *Implementation network of topology 1[2]*

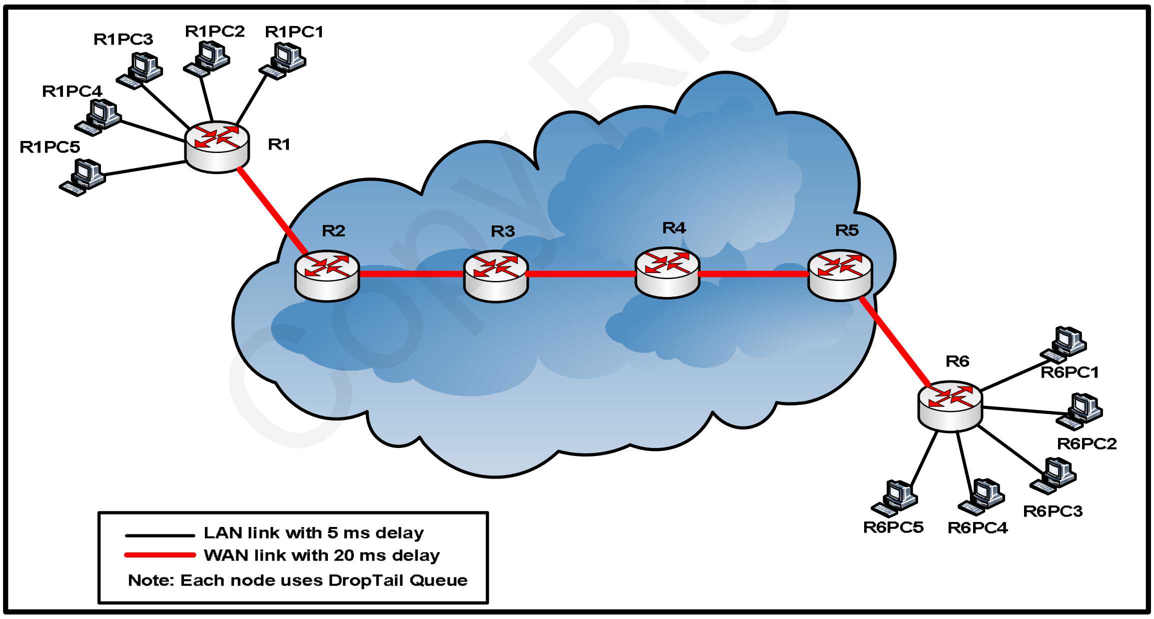

Fig. 6. *Implementation network of topology 2 [2]*

4. RESULTS AND DISCUSSION

4.1. Topology 1 Experiment 1: Number of Calls Supported

In this experiment, the number of concurrent calls that can be run at various bandwidths (100kbps, 150kbps, 200kbps, 250kbps, 300kbps and 350kbps) was explored. For each different link bandwidth, the number of concurrent calls was increased. When the packet loss starts the link is considered overloaded. Therefore, the number of concurrent calls for each link bandwidth is equivalent to the number of calls before the packet loss started. As we can from Figure 7, we can run more calls under the same channel bandwidth when using the ITTP protocol instead of IAX/UDP protocol. Obviously, this is because the ITTP protocol header size is less than the IAX/UDP protocol header size. Therefore, the ITTP protocol call consume less bandwidth than the IAX/UDP protocol call as shown in section 2.3.

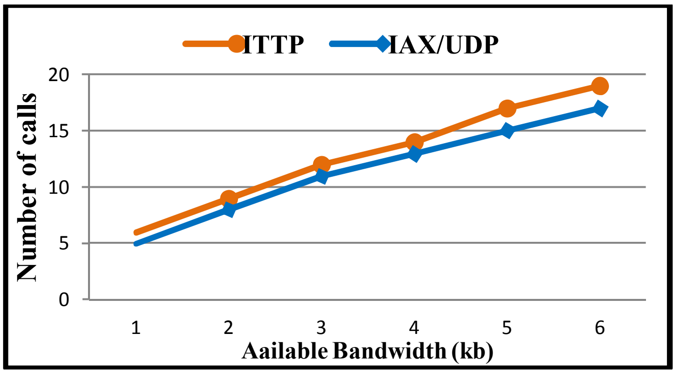

Fig. 7. *ITTP and IAX/UDP protocols number of calls supported*

4.2. Topology 1 Experiment 2: Goodput

In this experiment the goodput was used to evaluate the ITTP protocol compared to the IAX/UDP at various bandwidths (100kbps, 150kbps, 200kbps, 250kbps, 300kbps and 350kbps). Figure 8 depicts the goodput of the ITTP protocol and IAX/UDP protocols. As we can from Figure 7, the Goodput of the ITTP protocol is better the Goodput of the IAX/UDP protocols. Again, this is because the ITTP protocol header size is less than the IAX/UDP protocol header size. Therefore, more bandwidth is used to transfer the actual voice data when using the ITTP protocol than when using the IAX/UDP protocols.

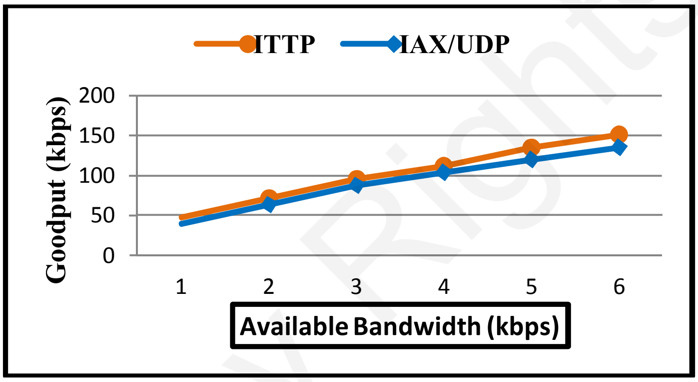

Fig. 8. *ITTP and IAX/UDP protocols Goodput*

4.3. Topology 2 Experiment 1: Packet loss

In this experiment, the packet loss of the ITTP protocol and the IAX/UDP protocols was investigated at various stream numbers between 1 and 5 streams. All the links have a bandwidth of 60 kbps. As we can see from Figure 9, the packet loss caused by the ITTP protocol is less than the packet loss caused by the IAX/UDP protocols under similar conditions. This is because the ITTP packet size is smaller than the IAX/UDP packet size, which allows more packet to be stored than the IAX/UDP before the buffer overflowed and decrease the packet loss. In addition, the difference of the packet size, make the packet processing of the ITTP faster than the packet processing of the IAX/UDP protocol [9] [10].Therefore, the router can process more ITTP protocol packets than IAX/UDP protocols packets, which decrease the packet loss as well.

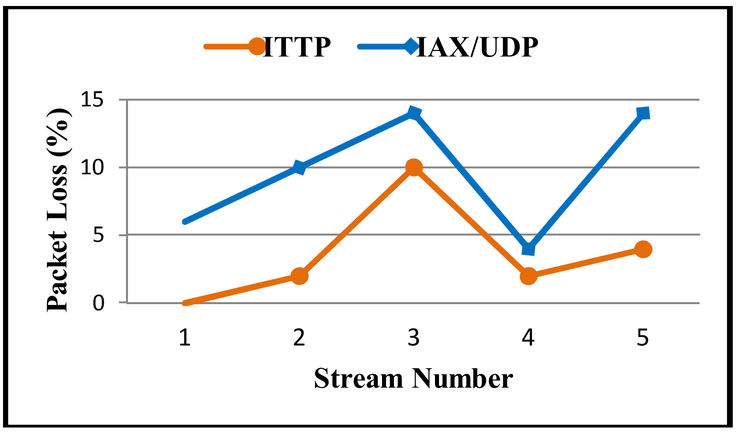

Fig. 9. *ITTP and IAX/UDP protocols packet loss ratio*

4.4. Topology 2 Experiment 2: Delay

In this experiment, two different delay cases of the ITTP protocol and the IAX/UDP protocols were investigated at various stream numbers between 1 and 5 streams. In the first case, all the links have a bandwidth of 104 kbps. In the second case, all the links have a bandwidth of 85kbps. As we can see in Figure 10 and 11, in the two cases, the ITTP protocol causes less delay than the IAX/UDP protocol. Whereas, the ITTP packet is less than the IAX/UDP packet which lead to: first, the transmission time required for the ITTP protocol packets is less than the transmission time required for the IAX/UDP protocol packets [10]. Second, the ITTP protocol packets will queue less time than the IAX/UDP protocol packets. Third, the IAX/UDP stream requires more bandwidth than the ITTP stream. Therefore, the number of IAX/UDP protocols packets queue in the buffer will be more than those for ITTP [10] [11].

The delay in the second case was more than the delay in the first case. This is because the 104kbps bandwidth in the first case is enough for 5 IAX/UDP protocols streams, while in the second case the 85kbps bandwidth is not. Therefore, in the second case, the queuing delay will be much more than the queuing delay in the first case. On the other hand, in the two cases the bandwidth is enough for 5 ITTP protocols streams. Therefore, the queuing delay was around zero.

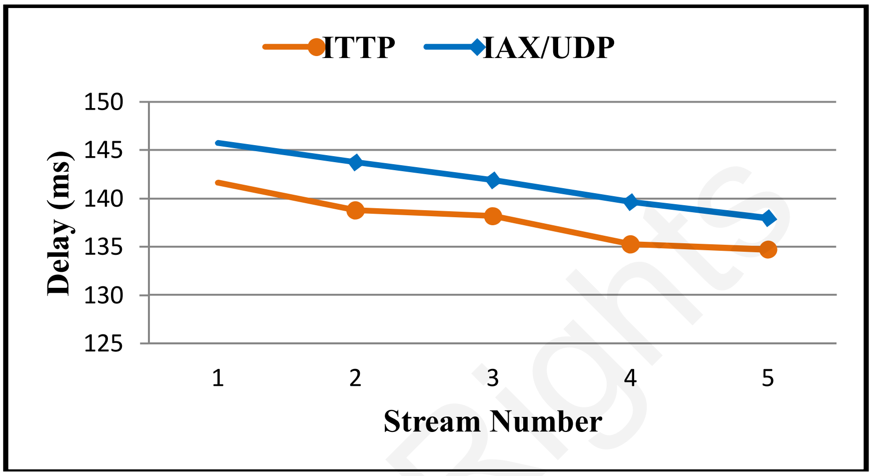

Fig. 10. *ITTP and IAX/UDP protocols delay, case 1*

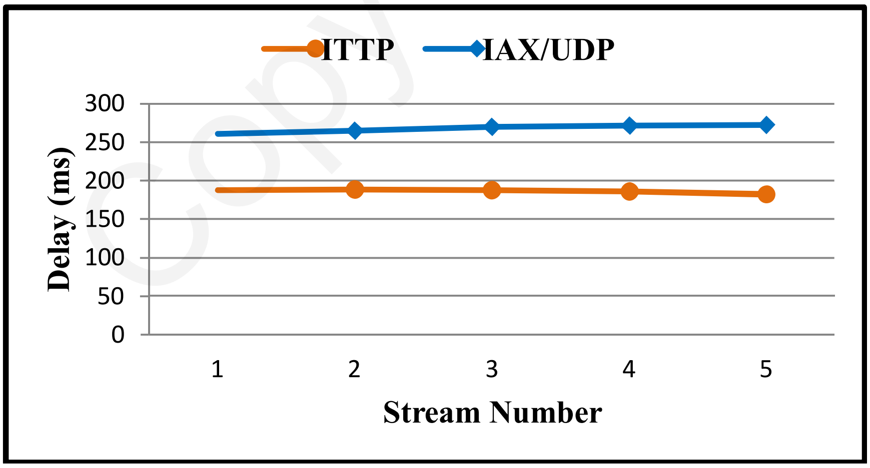

Fig. 11. *ITTP protocol and IAX/UDP protocols delay, case 2*

5. CONCLUSION

This paper explains the performance evaluation of the (ITTP) in comparison to IAX/UDP. Mathematical equations and implementation test have been used to demonstrate the ITTP performance and comparing it with the IAX/UDP protocol. The result showed that the ITTP is highly improved the bandwidth utilization efficiency compared to the IAX/UDP protocols. In addition, the ITTP protocol has improved the voice quality by decrease both the packet loss and the delay resulting from the IAX/UDP protocols. Therefore, the ITTP is a promising protocol to transport the VoIP applications data, shortening the problems resulting from the IAX/UDP protocols.

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