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Comparison of QoS Performance over WLAN, VoIP4 and VoIP6

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Abstract

VoIP stands for voice over internet protocol. It is one of the most widely used technologies. It enables users to send and transmit media over IP network. The transition from IPv4 to IPv6 provides many benefits for internet IPv6 is more efficient than IPv4. This paper presents a performance analysis of VoIP over WLAN using IPv4 and IPv6 and OPNET software program to simulate the protocols and to investigate the QoS parameters such as jitter, delay variation, packet send, packet received and throughputs for IP4 and IP6 and compare between them.

Keywords: *PSTN, VoIP, QoS, IP, OPNET, RTP, H.323.*

1. Introduction

Voice over IP (VoIP) is a technical method that is used to transmit data and voice over IP network. It allows all persons connected to the network to use this technique. Voice over IP enables sending the voice and data as digital packets through packet switched based network, instead of sending packet through Public Switched Telephone Network (PSTN) [1].

The first step for VOIP technique is converting the analog signal to digital signal, then compression and encoding the packets by using some codecs (G.711, G.722, etc). The following step is integration of the voice packet inside the data packet by through the real time protocol (RTP). Then forwarding packets through network in multi paths [2].

VoIP needs internet with high speed, high bandwidth plus hardware requirements such as:

- ATA (Analog Telephone Adapter): the ATA allows to connect a stander phone to computer, it takes the analog signal from traditional phone to convert it to digital data for transmission over Internet.
- IP Phone: These phones similar normal phones but instead of standard RJ-11 phone connectors, IP Phones have RJ-45 Ethernet connectors [1].

Voice over IP implemented by using protocols like (SIP, RTP, H.323).

- RTP (real time protocol): use this protocol to transmit any audio or video message between two computers, RTP is defined by the IETF in RFC 3550.
- H.323: consist of many standard protocols which improved by ITU, for transmission audio and data through network by used packet-based.
- SIP (Session Initiation Protocol): used to start, revise and end the call.

H.323 and SIP solve the problem of how two computers can initiate communication in order to exchange audio and video media streams, and allow users to do same things and establish communication [3].

Voice over IP uses the codices to compress and transmit the media through IP network. These codices are different according to bit space, frame length and the algorithm used.

G.711, G.723.1, G.726, G.728, G.729 [4].

IPv6 is an abbreviation to Internet protocol version 6, the extent of the computing machine and increased use of IP network and access to the Internet, require IP addressing to accessing the Internet. IPv4 has limited addresses compared with IPv6. It provides other technical benefits in addition to a larger addressing space content of 128 bytes compared with 23 bit in IPv4.

IPv6 has other benefits, it doesn't use the NAT (Network Address Translator). It has auto-configuration and built-in security and mobility. Also IPv6 has simpler header than IPv4 because it has fewer field. IPv4 has some issue not only in addresses also in services which it provide to customers and applications which need real time traffic. IPv6 is capable of solving these problems. IPv6 provides QoS for services and security.

IPv4 is 32 bytes, it can cover 4.3 billion addresses. The address represent about 192.168.0.3. Each colon starts from 0 to 255. In IPv4 there are 5 classes (A,B,C,D, reserved). Each class provides limited addresses for hosts and networks.

IPv6 is 128 bytes, it covers 340 trillion, trillion, trillion addresses, represented at Hexadecimal digits, each four digits separated by colon [5].

The header in IPv6 is not different totally from IPv4. IPv6 is 40 bytes while IPv4 is 20 bytes. IPv6 has fewer fields than IPv4, this means fast processing for data and high performance.

The version field determines the type of header (4 or 6) IPv4 or IPv6. The Traffic class field is 8 bytes from 0-7 and it determines priority of traffic. The flow label is 20 bits and provides the quality of service and the router provides the mechanism of processing for the traffic when it reaches this field. The Payload length is 16 bits and it determines the length of data and helps to transfer data up to 64 Kbyte. If the data exceeds 64K the extension header is used and it is capable of accommodating 4.3 million bytes. The type of extension header is to detect the type of next header. The Hop limit decreases after hop reach to zero. The source and destination are 16 bytes and they have the address [5].

Fast conversion from IPv4 to IPv6 is impossible because the IPv4 is a big network, and a lot of companies and organizations use IPv4 daily. It is possible to stop using IPv4 and replace it with IPv6, then the transition from IPv4 to IPv6 is very important due to the benefits and advantages of IPv6. This means the two versions must work together, the two protocols cannot understand each other to transfer data. The transition strategies are methods which provide a connection between IPv4 and IPv6 and they are:

- Dual-stack: the devices PC or router run both IPv4 and IPv6, the node will be able to receive all traffic.

- Tunnel: in tunneling, the two networks are used IPv4. IPv6 packets are tunneling across an IPv4 by encapsulating them in IPv4, this required the router configured with Dual-stack.

- Translation: this is similar with NAT, and change the header and payload from IPv4 to IPv6 with two methods: stateless and stateful.

Quality of service is very important especially for applications which need high performance like real time application. Quality of service is important if the network capacity is insufficient, especially for real-time multimedia applications such as voice over IP, online games and IP-TV, since these often require fixed byte rate and are delay sensitive. It is also important for networks where the capacity is a limited resource, for example in cellular data communication. Quality of service sometimes refers to the level of quality of service.

There are some important parameters in QoS:

Delay: the time which retard between the sending voice signal and the moment of arrival to destination, along time of each packet to arrive to destination, some time because queuing mechanism and routing direction in congestion.

Jitter: It is the variation of the delay in the voice packages that are delivered to destination. This variable time difference may determine interruptions in the voice signal.

Throughput: Throughput is a measure of how many of packets can be processed in a certain time. Throughput in a communication system is affected by several factors, including the limitations of underlying analog physical medium, available processing power of the system components, and end-user behavior.

Packet loss: the router may fail or lose the packets. The receiving application may ask for information that dropped to be retransmitted again, possibly causing severe delays in the overall transmission [4].

In [6] this paper presents properties of queuing mechanisms used to buffering signal of voice, by studying and analyzing the QoS parameters of each queuing mechanism by OPNET simulator. It concludes that PQ has better performance than WFQ in terms of amount of traffic and end – to – end delay. In [7] this paper provides an introduction to VoIP technology like: network structure, protocols, echo, delay and jitter.

In [8] this thesis provides possibilities to use network in building, as a model solution, the implementation of the VoIP network at the Technical University of Kosice and the Computer Networks Laboratory is described. In [9] this thesis provides encoding techniques that are based on the level of the security they provide and their usage now, G.711, G.72, G.729 by using OPNET.

In [10] this thesis presents problems related to Qos including system capacity planning, QoS parameter frication, run-time, resource allocation and load shedding. These algorithms and solutions form a framework for a DSMS to manage, control, deliver, and verify QoS requirements in a general DSMS.

In [11] this thesis distributed models for scalable QoS provisioning in the Internet, where network routers perform both data- plane and control- plane tasks without recourse to centralized, off-path control entities and proposes an architecture for the QoS subsystem of a next-generation, IP-based mobile telecommunications system, based on centralized control entities and designated QoS brokers.

The objective of this paper is to study, analyze, plan and design soft ware programs to Compare QoS performance of voice over WLAN over IPv4 and IPv6. Parameters which are considered in the performance are: WLAN delay, jitter, throughput, packet delay variation, Traffic received and Traffic send.

2. Descriptive Analysis

The network infrastructure is WLAN which consists of two different configurations with IPv4 and IPv6, the first scenario based on the IPv6 configuration, the second scenario based on IPv4 configuration, with the two access points connected to the switch, which connects to server and each access point has 7 workstations. In this network configuration with IPv6, the required parameters are measured by using OPNT simulator to get the results and show the effect the IPv6 & IPv4 on the performance QoS.

3. Simulation Parameters

Table 1: Simulation Environment

Parameters	Value
Topology	WLAN
IP technology	IPv6 & IPv4
Number of nodes	14
Network scale	Office
size	100*100 m^2
Link model	100 base full duplex
Technology	802.11 b
Data rate	11 Mbps
Codices used	G.711
Duration of simulation	10 minutes
Application	Voice over IP call (PCM)

4. Simulation

Descriptive analysis and simulation parameters are implemented by using OPNET software program 14.5 to get the results, as shown in figure 1:

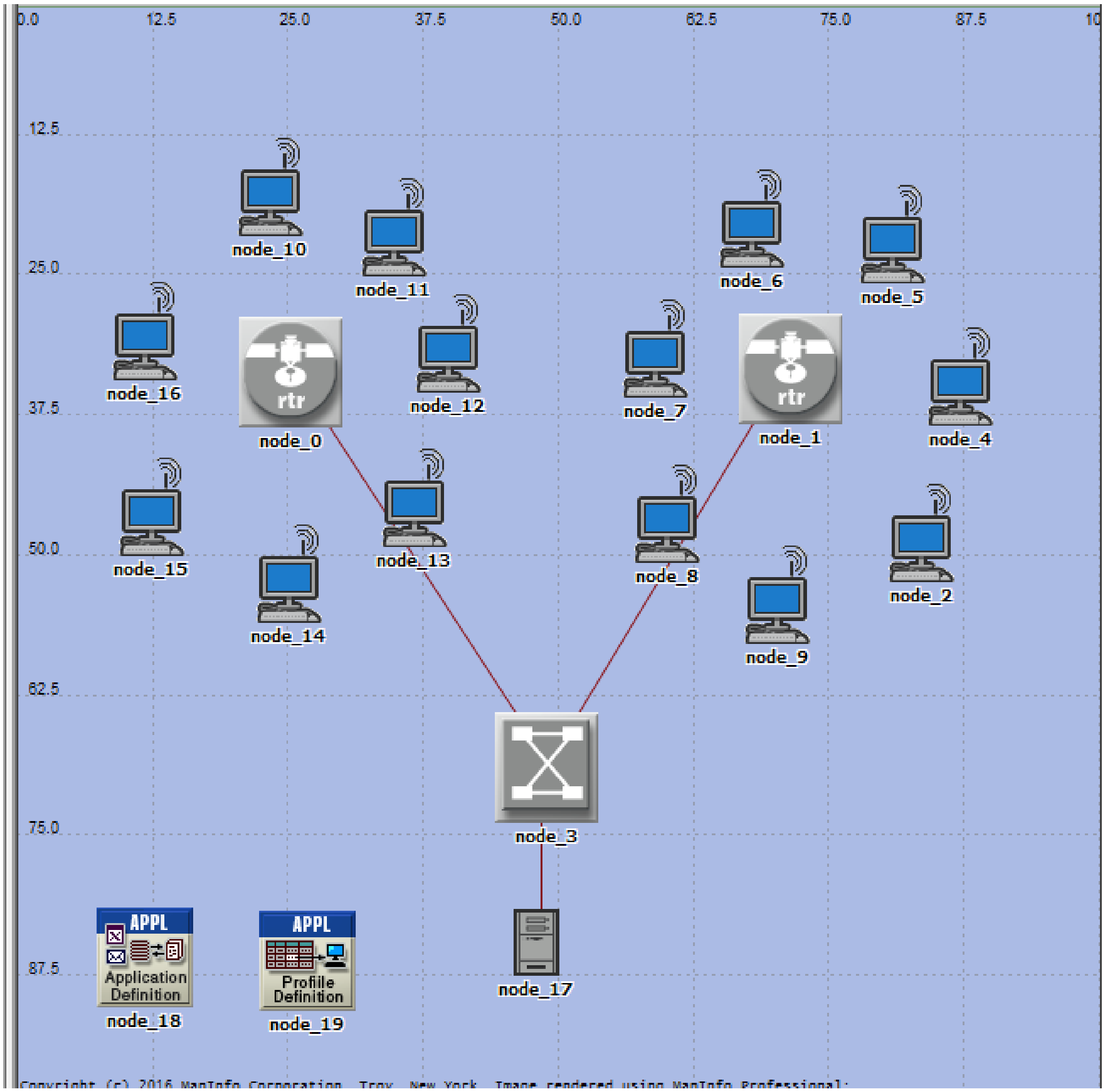

Fig 1: Network Topology

5. Results

The simulation was implemented to get the results for the parameters of QoS.

1. Jitter

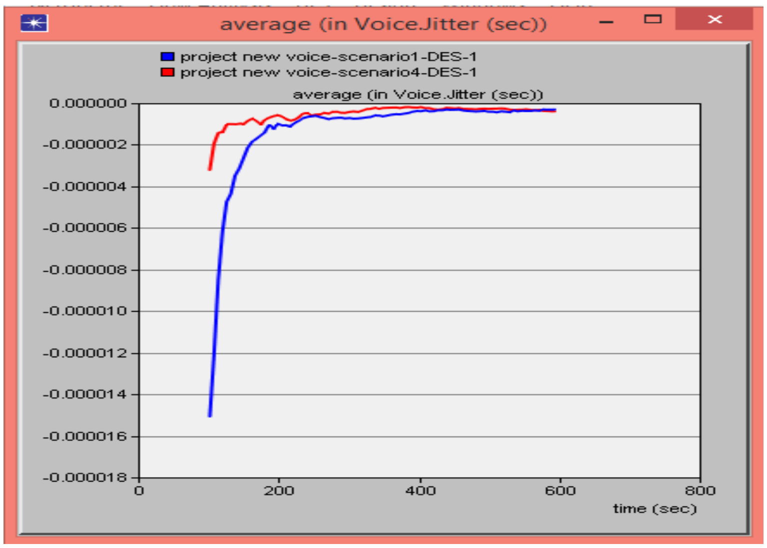

Fig 2: Jitter (scenario 1: IPv6, scenario 4: IPv4)

2. Packet Delay Variation

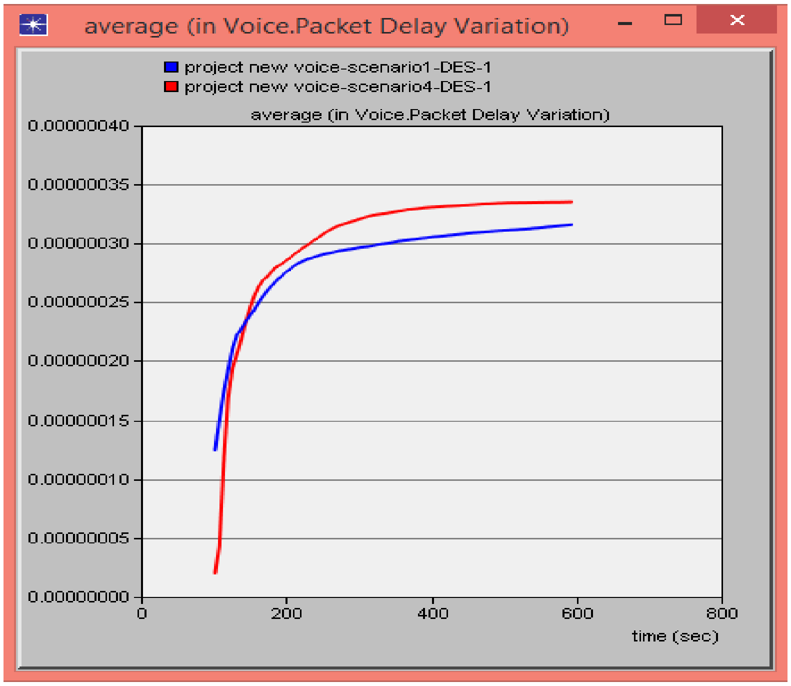

Fig 3: packet delay variation (scenario 1: IPv6, scenario 4: IPv4)

3. Traffic Sent(Packet/Sec)

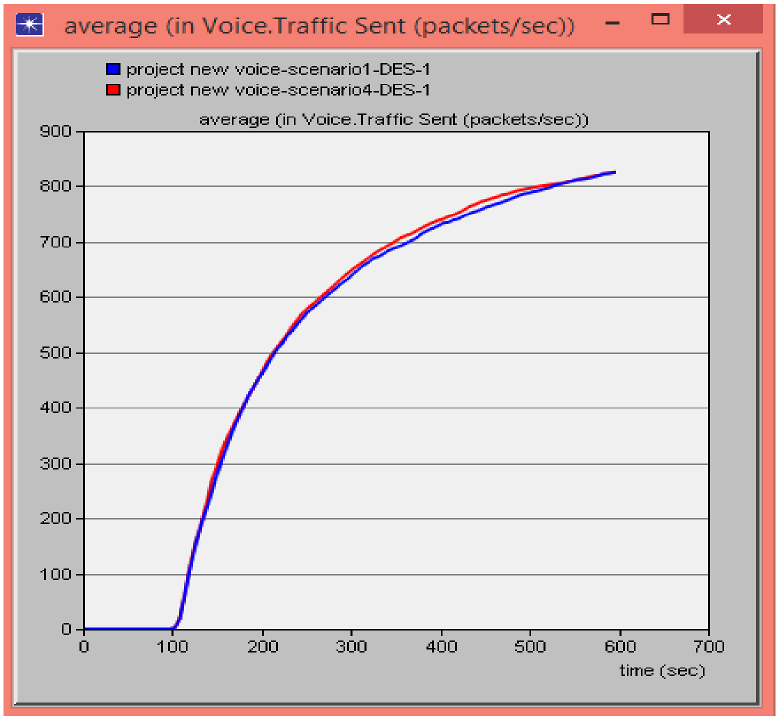

Fig 4: Traffic sent (scenario 1: IPv6, scenario 4: IPv4)

4. Traffic Received

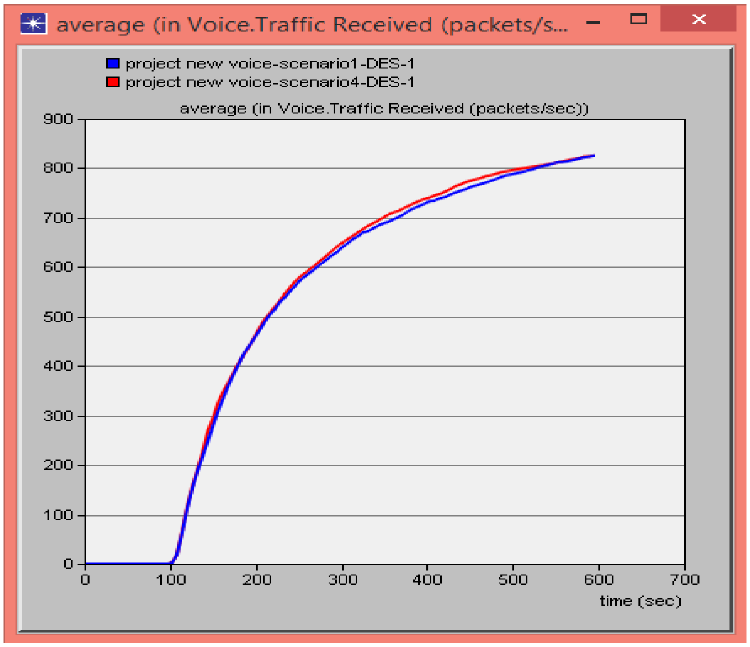

Fig 5: Traffic received (scenario 1: IPv6, scenario 4: IPv4)

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5. Wireless LAN Delay:

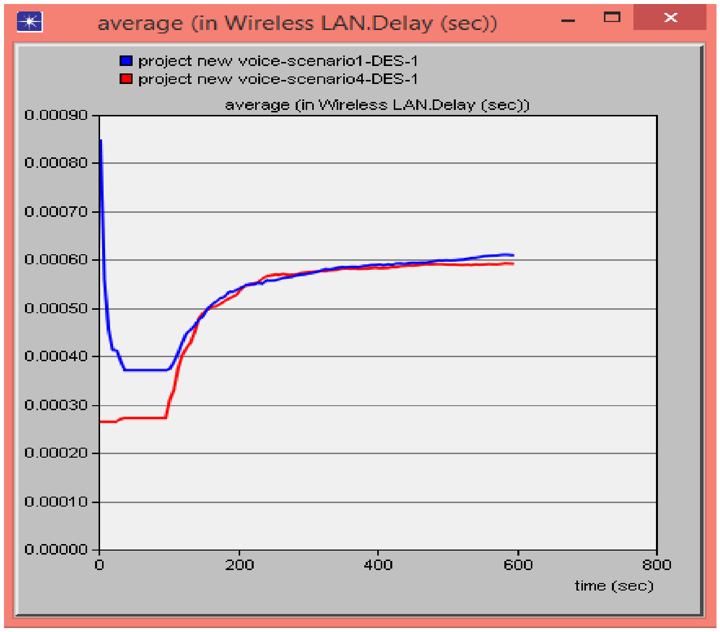

Fig 6: Wireless LAN Delay (scenario 1: IPv6, scenario 4: IPv4)

6. Throughput

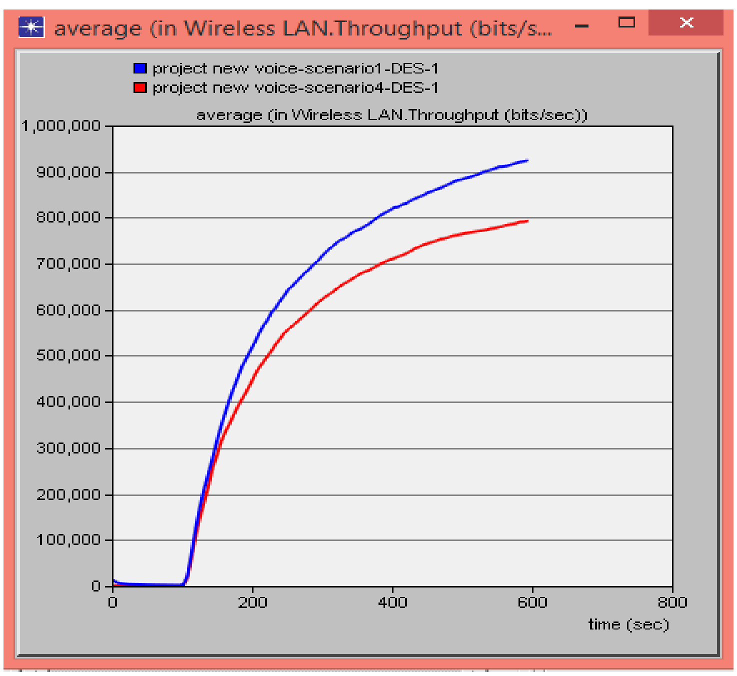

Fig 7: Throughput (Scenario 1: IPv6, Scenario 4: IPv4)

6. Results Analysis

From all previous graphs notice that, the jitter in IPv6 increase to value (-3.037) also IPv4 has similar value. In traffic sent in IPv6 and IPv4 have a similar value. In traffic received in IPv6 and IPv4 have a similar value. In wireless LAN delay IPv6 decreases then increases to value (0.000609) but in IPv4 the value increases to (0.00050). In throughput in IPv6 is better than IPv4.

Table 2: Values of Parameters

Parameters	expected value	variance	standard deviation
Jitter IPv6	-1.2858	5.5043	2.3461
Jitter IPv4	-5.0937	1.89493	4.35308
MOS value IPv6	3.6919	0.00	0.00
MOS value IPv4			
Packet delay variation IPv6	0.0606	1.6493	4.0611
Traffic sent IPv6	509.647	89,844.3	299.7404
Traffic sent IPv4	515.77	91,319.3	302.1909
Traffic received IPv6	509.645	89,844.2	299.7402
Traffic received IPv4	515.77	91,319.2	302.1906
Wireless LAN Delay IPv6	0.00053	7.395982	8.59998
Wireless LAN Delay IPv4	0.00050	1.434428	0.000119
Throughput IPv6	571,946.7	112,037,8	334,720.5
Throughput IPv4	495,286.2	84,085,4	289,974.

7. Conclusion

In this paper, the goal of study and objectives were achieved by using OPNET simulator to analyze the characteristics of codices which used to compress and transmit the voice through IP network. It is clear that , the characteristics of voice over IPv6 & IPv4 is not too different in these scenarios, but IPv6 is still better than IPv4 in transmission of the voice and less affected by the Qos parameters, while we notice the wireless LAN delay Throughput,

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