VOICE OVER IP VERSION 6 PERFORMANCE

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Abstract:

Voice over IP (VoIP) is a technologies to transmit the voice and data through IP network instead of sending the packet through Public Switched Telephone Network (PSTN), it offer many of benefits. The advance of technology requires the Transition from IPv4 to IPv6 which increase the ability and performance of the network. The objective of this paper are tostudy and analyze the characteristic of voice codecs to evaluate performance of voice over IPv6, using OPNET simulator to determine the effect of these codecs in the QoS parameters.

Keywords: PSTN, IPv4, IPv6, OPNET, G.711 and G.723.1

1. **Introduction:**

VOIP is an abbreviation for voice over Internet protocol, VOIP is a transmission of voice through network using Internet protocol (IP), this technology enables the transfer of voice, Conferences and multimedia over internet protocol. VOIP is the easiest way to make a call by sending packet through packet switched based network, instead of sending packet through Public Switched Telephone Network (PSTN) [1].

In this technique, the analog signal is convertor to digital signal which is divided to packets, this packets uses Internet protocol to arrive to destination in several paths. Voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network [2].

VoIP technique uses hard ware requirements as Internet service with high bandwidth and high speed, ATA (Analog Telephone Adapter) the ATA allows user to connect a stander phone to computer, the ATA is an analog-to-digital convertor, IP Phone with RJ-45 Ethernet connectors instead of RJ-11 phone connectors [1].

Voice over IP is implemented by using protocols like

VOIP signaling protocols: 1-Session Control Protocols: is responsible for the establishment, drooping, and keeping of call. 2-Media Control Protocols: is responsible for creating and dropping the media connections, used to control flows media through gateway.

Real time protocols (RTP): is responsible for applications that have real time characteristics, deliver and manage multimedia through IP network [3].

Codecs is converts an audio signal into compressed digital for transmission, to transmit the voice through IP network used codecs to compress and transmit the media through IP network. These codecs are different according to bit of rate, frame length and the algorithm used, like G.711, G.723.1, G.726, G.728, G.729 [4].

Internet protocol version 6 (IPv6) is improvement for (IPv4), IPv6 comes to complete, enhance and Extend IPv4 works, every devices laptop, desktop and all computing machines on the Internet assigned with IP address to access Internet. These devices will be increased and IPv4 addresses have some inability with rapid development of the Internet. IPv6 has very large address space content of 128 bit compared with 23 bit in IPv4. That means IPv6 can cover 340 trillion, trillion, trillion nodes whereas IPv4 is only capable of 4.3 billion nodes[5] .

Quality of service (QoS): is an expression refers to performance of telephony and computer networks, that provide better service, and ability to give the different priority for different application, users, data traffic, and guarantee certain level of services, such as bit error, jitter, delay, throughput, packet loss and bit rate.

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: is the variation of delay in the voice packages that are delivered to destination. This variable time difference may determine interruptions in the voice signal [4].

In [6, 7] the authors presentproperties, performance and analysis of VoIP using IPv4 & IPv6 and investigates areas of performance weakness, they concluded that the transition from IPv4 to IPv6 is present significant improvements for VoIP.

In [8, 9] the authors examine the quality parameters related to VoIP, the two quality parameters conceder are delay and delay variance and the characteristics of VoIP traffic in SIP based soft Phone system is analysis.

In [10, 11] the author definition cross-layer approach to system design derives from enabling interaction among protocols operating at different layers of the protocol stack in order to provide improvement in terms of some performance metric, cross-layer approach to system design derives from enabling interaction among protocols operating at different layers of the protocol stack in order to provide improvement in terms of some performance metric. And discussed problems and issues related to QoS support in a general DSMS, present algorithms and solutions form a framework for a DSMS to manage, control, deliver, and verify QoS requirements in a general DSMS.

The objective of this paper is to study, analysis, plan and design soft were program to evaluate performance of voice over IP version 6, parameters which are consideration in the performance are: delay, jitter, and throughput and packet loss.

1. **Descriptive analysis:**

The network infrastructure is WLAN consists of three scenarios, each scenarios consists of two access points connected to the switch, which connects to server, the network has 7 work-stations for each access point, IPv6 is applied.

OPNET 14.5 was used to simulate three scenarios, each scenario has different configuration to exam effect of codecs G.711, G.729A and G.723.1 sequentially in performance VoIPv6.

1. **simulation environment:**

Parameters	Value
Topology	WLAN
IP technology	IPv6
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
Codecs used	G.711, G.729A and G.723.1
Duration of simulation	10 minutes
Application	Voice over IP call (PCM)

Table 1: simulation environment.

2. simulation:

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

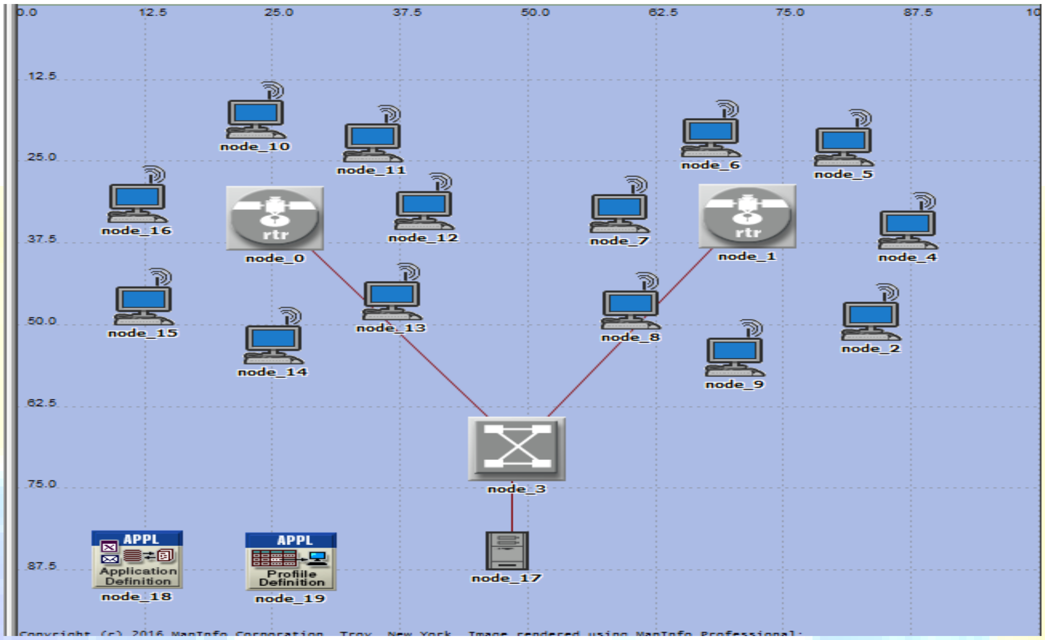

Fig (1): Network topology.

3. Results:

The results of implementation the scenarios are:

1. Jitter:

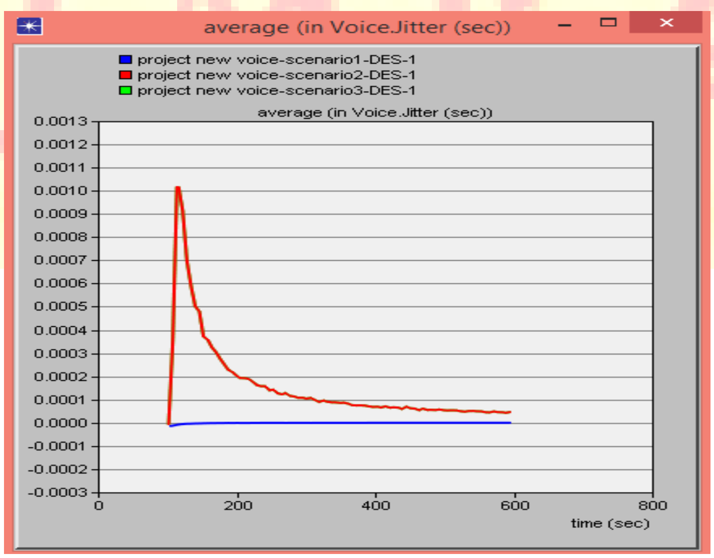

Fig (2): value of Jitter.

(Scenario 1: G.711, Scenario 2: G.729.A, Scenario 3: G723.1)

2. Packet delay variation:

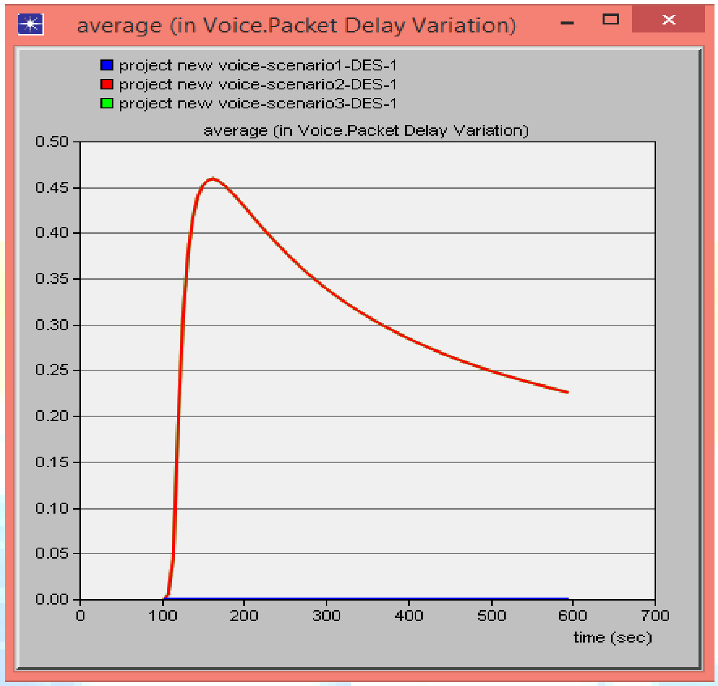

Fig (3): Packet delay variation.

(Scenario 1: G.711, Scenario 2: G.729.A, Scenario 3: G723.1)

3. Wireless LAN delay:

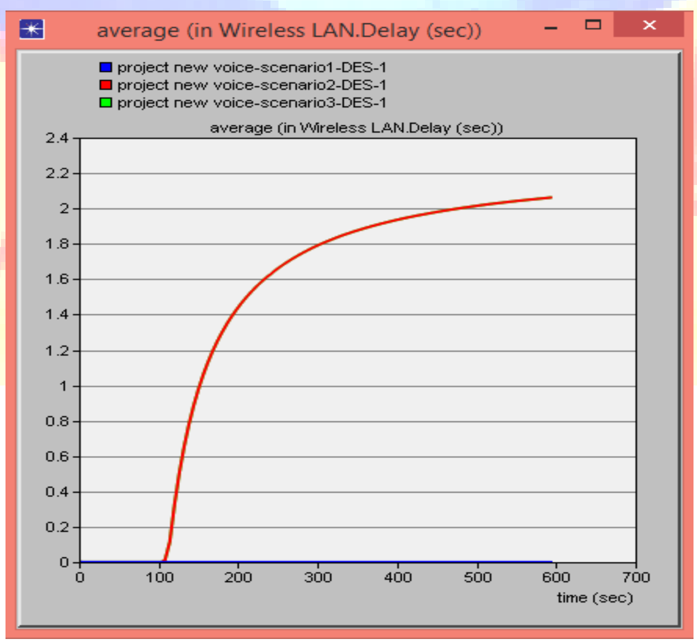

Fig (4): Wireless LAN delay.

(Scenario 1: G.711, Scenario 2: G.729.A, Scenario 3: G723.1)

4. Wireless LAN Throughput:

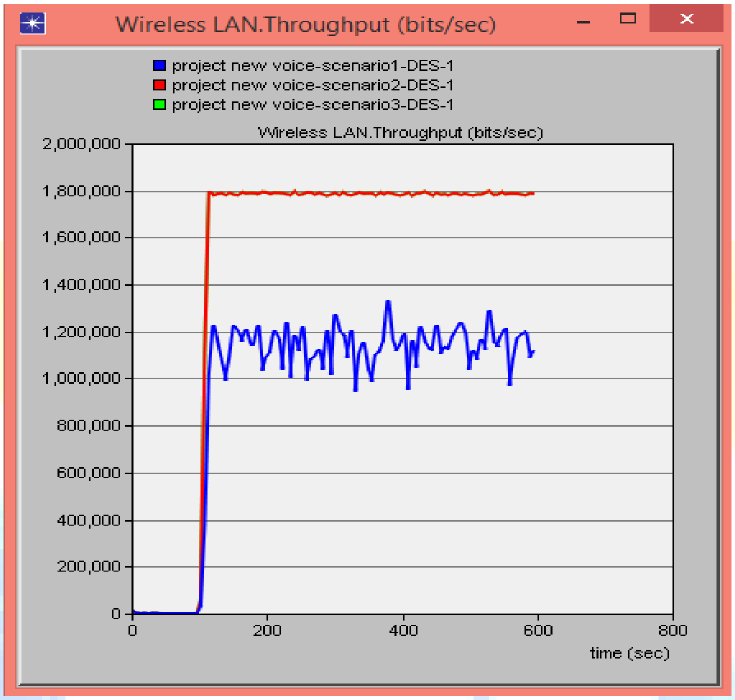

Fig (5): Wireless LAN Throughput.

(Scenario 1: G.711, Scenario 2: G.729.A, Scenario 3: G723.1)

4. Results analysis:

From all previous graphs, notices that: the Jitter for codecs was shows that G.711 is less effect by jitter start value from (0.00) to end value in (-3.03) where G.729.A, G.723.1 had higher effect by jitter, start values in (0.00), (0.00) then increase to end values in (4.61) ,(4.61) respectively.

The Packet delay variation for codecs showed that G.711 is less effect by jitter start value from (0.00) to end value in (3.15) where G.729.A, G.723.1 had higher effect by jitter, start values in (0.00), (0.00) then increase to end values in (0.2257) ,(0.2257) respectively.

The Wireless LAN delay for codecs showed that G.711 is less effect by jitter start value from (0.0008) to end value in (0.0006) where G.729.A, G.723.1 had higher effect by jitter, start values in (0.0008), (0.0008) then increase to end values in (2.0621) ,(2.0621) respectively.

The Wireless LAN Throughput for codecs showed that G.711 is less effect by jitter start value from (12,618.6) to end value in (924,725.76) where G.729.A, G.723.1 had higher effect by jitter, start values in (12,618.6), (12,618.6) then increase to end values in (1,459,110.0) ,(1,459,110.0) respectively.

Parameters	expected value	variance	standard deviation
Jitter (G.7.11)	-1.2858	5.5043	2.3461
Jitter (G.7.29.A)	0.0001	3.2786	0.00018
Jitter (G.7.23.1)	0.0001	3.2786	0.00018
Packet delay variation (G.711)	2.87549	1.3789	3.71339
Packet delay variation (G.729.A)	0.3071	0.0083	0.0915
Packet delay variation (G.723.1)	0.3071	0.0083	0.0915
Wireless LAN delay (G.711)	0.000538	7.395982	8.599989
Wireless LAN delay (G.729.A)	1.38850	0.59586	0.77192
Wireless LAN delay (G.723.1)	1.38850	0.59586	0.77192
Wireless LAN Throughput (G.711)	571,946.72	112,037,8	334,720.5
Wireless LAN Throughput (G.729.A)	911,807.564	279,310,986	528,498.80
Wireless LAN Throughput (G.723.1)	911,807.564	279,310,986	528,498.80

Conclusion

In this paper, the goal of study and objectives were achieved by using OPNET simulator to analysis the characteristics of codecs which usestocompress and transmit the voice through IP network, observed that the code G7.11 is better performance than G.729.A and G7.23.1 in Jitter, Delay variation, Wireless LAN Delay and Wireless LAN Throughput. The codecs G.729.A and G.723.1 have the smaller characteristics. That means the code G.711 better when data transmission as real time application.

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