Performance Analysis Of Active Queue Management (AQM) In VOIP Using Different Voice Encoder Scheme

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Abstract: Voice over Internet Protocol (VoIP) is a rapidly growing technology that enables transport of voice over data networks such as Ethernet Wide area networks (WANs) due to this important different codec scheme is developed to meet the QoS requirements. This thesis presents a comprehensive study about the impact of active queue management (AQM) on Voice over Internet Protocol (VoIP) quality of service using different codec scheme such as G711, G723, G729 and GSM using simulations tools. The evaluation is done using the OPNET Modeler, which provides a convenient and easy-to-use platform for simulating large scale networks and this also give a power to go through different levels of designing a network even with the ability to program the mechanism you want which is used here to implement two types of AQM mechanism which is not included by default in the OPNET and these two mechanisms are ARED and GRED. The performance metrics used in the study are jitter, throughput and delay. The study shows that G.711 and G729 codecs in a simulation gives a significant result for the performance of VoIP that codec G711 and G.729A has acceptable throughput and less deviation of received to transmit packet as compared to GSM and G.723 also average delay like end to end delay and Voice jitter is lesser in codec G711 and G.729 as compared to the other two referenced codecs.

Keywords: VOIP,AQM,PQM,RED,ARED,GRED,WRED, QoS, ToS.

1.1 Introduction

The Computer network, and well known computer network is the internet become one of the main things that daily used because has an enormous impact in our society, there is competition between users or members of the society to get reliable and high speed in connection to enjoy internet services, but every competition have challenge that shows in form of situation called congestion ,congestion occurs due to the increase in numbers of users so that a huge amount of data been transmitted and received from one node to another in matter of second or less and sometimes it be difficult for network to control , and this leads network to suffer from congestion that needs to be controlled, so what is the congestion ?? Congestion is the situation when too many sources sending too much data too fast for the network to handle, and this affect the quality of service (QoS) because of the increase of the delay, jitter and the packet loss, therefore it is important to reduce this problem as much as possible. There are different approaches for dealing with congestions: passive queue management (PQM) and active queue management (AQM) and others. Drop Tail is a representative PQM algorithm which only sets a maximum length for each queue at the router, a phenomenon in which all senders sharing the same bottleneck router/link shut down their transmission windows at almost the same time and this make the global synchronization problem. To solve this problem AQM techniques is used trying to detect congestion before it happen by using some mathematical model and algorithms and make the management of the queue become active and avoid consequences such as queuing delay and packet loss [1]

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1.2 Voice over Internet Protocol (VoIP)

Despite the growing popularity of data services, voice services still remain the largest income source for network service providers. The two most popular ways of providing voice services are packet switched telephone networks (PSTNs) and wireless cellular networks. The deployment of both of these forms of networks requires infrastructures that are usually very expensive. Alternative solutions are being sought which can deliver good-quality voice services at a relatively lower cost. One way to achieve low cost is to use the already existing IP infra-structure. Means that carry voice with computer data in the same links, Protocols that are used to carry voice signals over the IP network are commonly referred to as voice-over-IP (VoIP) protocols [2]. VoIP application typically works as follows. First, a voice signal is sampled, digitized, and encoded using a given algorithm/coder. The encoded data (called frames) is packetized and transmitted using RTP/UDP/IP. At the receiver's side, data is de-packetized and forwarded to a play out buffer, which smoothest out the delay incurred in the network. Finally, the data is decoded and the voice signal is reconstructed. A wide range of impairments are taken account such as codec impairments, end to end delay, jitter, and packet loss, as well as noise and echo.in VoIP voice signal is compressed and converted to digital voice packets, VoIP then uses the Internet Protocol (IP) for managing voice packets over IP network. Therefore, VoIP can be deployed on any IP enabled data network, such as the Internet, Ethernet, fabric or wireless network [3]. There are several different types of VoIP service depending on the infrastructure used for the communication: computer to computer based VoIP (VoIP device to another VoIP device), computer to Phone based VoIP (VoIP device to a PSTN device), and Phone to Phone based VoIP (PSTN device to another PSTN device) [4]. Each type of them has different set of requirements in additional to the boundaries on the technical level crucial services such as VOIP have to be available and perform in an acceptable quality, these circumstances require efficient utilization of scare bandwidth.

1.3.1 VoIP Compression Algorithms

Codecs generally provide a compression capability to save network bandwidth. Currently, there are many different audio codecs available for voice applications. The simplest and most widely used codecs are G.711, G.723 and G.729 [7]. The simplest encoder scheme is G.711 (64 kb/s). G.711 is the sample based which uses Pulse Code Modulation (PCM). The acceptable packet loss factor of G.711 is up to 0.928%.

1.3.2 RED: Random Early Detection

Sally Floyd and Van Jacobson present in [6] a mechanism called Random Early Detection (RED) that aims congestion avoidance. Their work is motivated by the goal to keep average queue sizes in routers small. This is done by dropping or marking some packets that have a position in the queue exceeding a certain threshold. For marking packets ECN can be used, indicating congestion on the route. System-wide parameters Qmin and Qmax define the threshold boundaries of the queue size. Upon arrival of a new packet in the system the average queue size Qavg of the flow is calculated and compared to Qmin and Qmax. Qavg is updated each time a packet arrives with following formula:

$$Qavg = (1 - Wq) \cdot Qavg + Wq \cdot Qcurr, \qquad \textbf{(1.1)}$$

Where Wq is the weight of the former average queue size and Qcurr the current queue size When Qavg exceeds Qmax the packet is marked. In the case that Qavg is within the boundaries of Qmin and Qmax, the marking probability Pm is calculated using following equation:

$$pm = pa\ avarage \div (1 - count.pavag) \qquad \textbf{(1.2)}$$

Where count is the number of packets since the last marked packet and Pavg is defined as followed:

$$Pavag = Pm \max.((Qavag - Qmin) \div (Qmax - Qmin)) \qquad \textbf{(1.3)}$$

The packet is marked with a probability of Pm, in this case count is reset. If the packet is not marked, count is incremented. With this mechanism the average queue size can be controlled and congestion can be avoided. With the parameter Wq additionally the burst-awareness of the mechanism can be modeled. Qmin and Qmax define the expected range of queue length, Qmin defines the minimum queue length at which no packets are dropped, as Qavg exceeds Qmin the dropping probability increases with increasing Qavg and count, up to the maximum dropping probability Pmax avg. These parameters can be configured to suit to different environments. Fairness is provided based on the assumption that the number of dropped packets correlates with the utilization of the bandwidth by the specific flow [6]. In [7] the authors mention the goals that they met after using RED and this is summarize for the following:

- Congestion Avoidance
- Simplicity
- Appropriate time scales
- No global synchronization
- Fairness

And they summarize the limitation that One of RED's main weaknesses is that the average queue size varies with the level of congestion and with the parameter settings means that the average queuing delay from RED is sensitive to the traffic load and to parameters, and is therefore not predictable in advance.

1.3.2 ARED Adaptive Random Early Detection

Floyd, Gummadi and Shenker present in [8] a slight modification of RED, called Adaptive RED that provides in face of congestion average pre-defined delay to the flows. The strategy Adaptive RED applies is adapting the parameters of RED to the current situation. The parameters are Qmin and Qmax being lower and upper threshold of expected queue length, Wq is the weight of the current queue length for the calculation of the average queue length. Finally Pmaxm is the maximum probability for marking packets that a system can achieve. The authors argue that high link utilization, requiring large buffers, and low transfer delays, requiring small buffers, are concurrent aims and a trade-off has to be found. RED randomly drops packets with probability related to the current average queue size. Only congestion and the parameter setting have effect on the balance between link utilization and low delays.

1.3.3 GRED Gentle Random Early Detection

To deal with the above RED's problem, GRED was proposed as a variant of RED, in which GRED enhances the way of tuning some of RED's parameters such as max threshold and max D [9]. Moreover, GRED maintains the aql value at a level on a router buffer which is between the min threshold and max threshold positions, this level is named aql T [9]. Maintaining aql value at aql T level avoids increasing the router buffer size to be above the max threshold position, and therefore smaller number of packets is dropped. GRED was proposed as an AQM method that controls congestion at router buffers preliminary [10]. GRED aims to solve some of RED's problems [9, 8,10] explained in Section 1.4.1

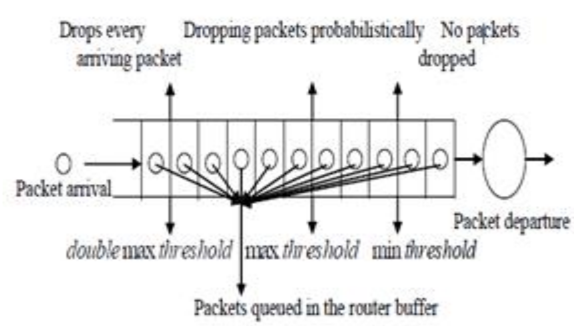


Figure 1.1 *Router buffer in GRED*

In [11] comparison's conclusion is presented as follows: GRED show improved result and solve many of RED problems mentioned earlier in this section

1.3.4 WRED Weighted Random Early Detection
By randomly dropping packets prior to periods of high congestion, WRED tells the packet source to decrease its transmission rate. If the packet source is using TCP, it will decrease its transmission rate until all the packets reach their destination, which indicates that the congestion is cleared. WRED generally drops packets selectively based on IP precedence. Packets with a higher IP precedence are less likely to be dropped than packets with a lower precedence. Thus, the higher the priority of a packet, the higher the probability that the packet will be delivered. WRED is only useful when the bulk of the traffic is TCP/IP traffic. With TCP, dropped packets indicate congestion, so the packet source will reduce its transmission rate. With other protocols, packet sources may not respond or may resend dropped packets at the same rate. Thus, dropping packets does not decrease congestion. WRED treats non-IP traffic as precedence 0, the lowest precedence. Therefore, non-IP traffic, in general, is more likely to be dropped than IP traffic, Figure1.2 illustrates how WRED works.

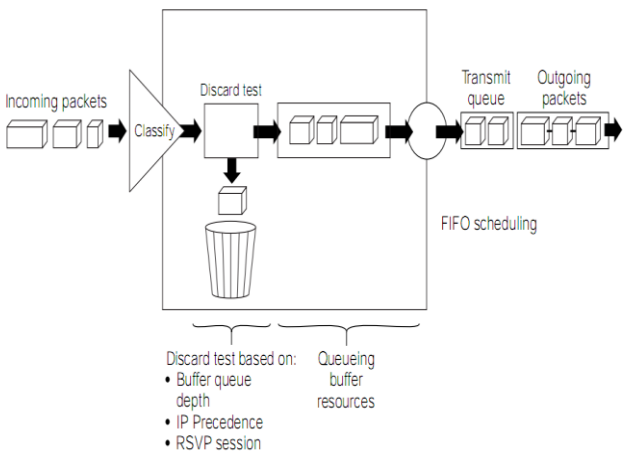


Figure 1.2 *Weighted Random Early Detection*

2.2 VOIP CODECS
A codec is the term used for the word coder-decoder, converts of analog audio signals into compressed digital form for transmission and then back into an uncompressed audio signal for the reception. There are different codec types based on the selected sampling rate, data rate, and implemented compression. The most common codecs used for VoIP applications are G.711, G729A, G723.1, GSM etc. each of which varies in the sound quality, the bandwidth required, the computational requirements, encoding algorithm and coding delay [3, 9, and 10]: In our evaluations codecs of ITU standards for audio compression and decompression are used. [5]

2.3 Traffic characteristics in VoIP
VoIP exhibits certain properties that affect the quality of voice delivery over a network. These properties primarily include delay, jitter and loss. Delay is the time taken for a packet to travel from source to destination. QoS is assured in VoIP when delay is less than or equal to 150ms. However, it is still acceptable if delay is between 150ms and 400ms. Delay can be reduced by using appropriate codec and queuing algorithms. It can also be reduced by sending fragmented packets over the network and reassembling them at the destination. This however increases the overhead load on the network device. Jitter in VoIP is a variation in delay. It is the difference between the minimum and the maximum end-to-end delay and signifies the variable delay within the network. Packets transmitted at constant rate are expected to be received at constant rate. However, due to network conditions packets may arrive at variable rate. To balance jitter, a jitter-buffer is used to tailor out all packets received at irregular intervals and shape them to constant intervals before the packets are processed. Apart from delay and jitter, packet loss and error also affects the quality of voice delivery. Packet loss is primarily due to buffer over flow within the router`s memory but could also be as a result of bad transmission, late delivery or general network errors. Unlike in data applications where lost or error in information can be retransmitted, in VoIP retransmission delay is unacceptable and therefore traffic engineering policies has to be implemented to limit loss to a minimal level. In this context, scheduling algorithms have an important role to prioritize and forward the real time traffic over traditional data traffic [12]

2.4 VoIP Application Attribute
The experiments are based on G.711, G.729, G723.1 and GSM encoder. The G.711 encoder is based on uncompressed Pulse Code Modulation (PCM) voice and could be used to encode straight from a traditional telephone network, [13]. The G.711 use 8 bits per sample and each sample is generated every 125 microseconds with the use of PCM and this leads to a bit rate of 64 Kbps. The G723.1 use Multi-rate Coder and this compressed voice and leads to a bit rate of (5.3 and 6.4) kbps. The GSM encoder use RPE-LTP (Regular Pulse Excitation Long-Term this leads 13 kbps. Prediction [14] The G.729 generates compressed voice and typically operates at 8kbps and is ideal for VoIP because of its low bandwidth Voice speech consists of talk spurt length and silence length which have default values. Talk spurt is defined in VoIP as the length of sound in-between silence period. The Type of Service (ToS) field value is set to interactive voice. The G.711 voice frame used in OPNET is 32 bytes long and therefore of 4 milliseconds duration. And table below show the VoIP Codec Protocols and Algorithms:

Table 2.3.1: *VoIP Codec Protocols and Algorithms*

Codec	Algorithm	kbps
G.711	PCM (Pulse Code Modulation)	64
G.723	Multi-rate Coder	5.3 and 6.4
G.729	CS-ACELP (Conjugate Structure Algebraic-Code Excited Linear	8

	Prediction)	
GSM	RPE-LTP (Regular Pulse Excitation Long-Term Prediction)	13

3.4.1 Topology description

A typical topology used in this project is in the form of "bottleneck", with five users and one Ethernet server, two switches and two routers. The capacity of the link between the two routers will be much lower compared to the other links, to create the situation of congestion control and experience packet drop and other effects. Three other important configuration parameters were used: application, profile and QoS profiles.

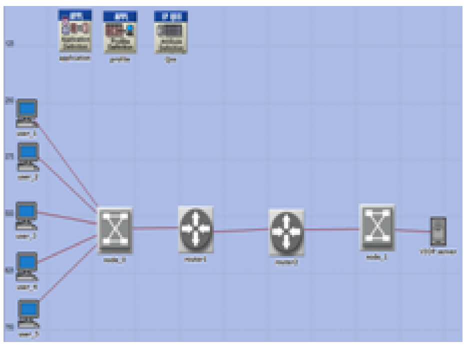


Figure 3.1 *Simple dumbbell Topology*

4.1 Simulation parameters:

A detailed simulation network modelling were described. In order to obtain graphical results, before running simulation, a number of statistics in OPNET need to be configured for VoIP network components include VoIP traffic, switches, router, client, and links and environments configurations. The duration of OPNET simulation was set to 30 minutes (duration time for logical network). The VoIP traffic started at 60 seconds after the simulation is initially started. Every simulation stops at 8 minutes, the statistical and graphical results are generated by OPNET. The below table shows the simulation parameters that used in this scenario, this scenario based on changing the number of users with different codec scheme algorithm while the other parameters remain constant

Table (4.1) Simulation parameters

Number of User	300
Data forwarding rate (packets/second)	500000
Maximum queue size	100
Minimum threshold	10
Maximum threshold	30
Mark probability denominator	0.1
Simulation duration	30 min
Encoder scheme	G.711 / G.723 / G.729 GSM
Voice frame per packet	5
Type of service	Interactive voice (6)

4.1.4 Number of client (300):

4.1.4.1 Voice jitter results:

The figures 4.13 shows the jitter result when using four different active queue management, after we change the voice codec scheme in the 4.13 ARED, GRED, RED, WRED shows same jitter results in part (a), in the part (b) case we found that WRED get the minimum jitter and ARED, GRED, RED get the maximum value of the jitter, in the part (c) ARED, RED, GRED,WRED get same jitter results, in the part (d) we found that WRED get the minimum value of jitter and RED, ARED, GRED get the maximum value of the jitter.

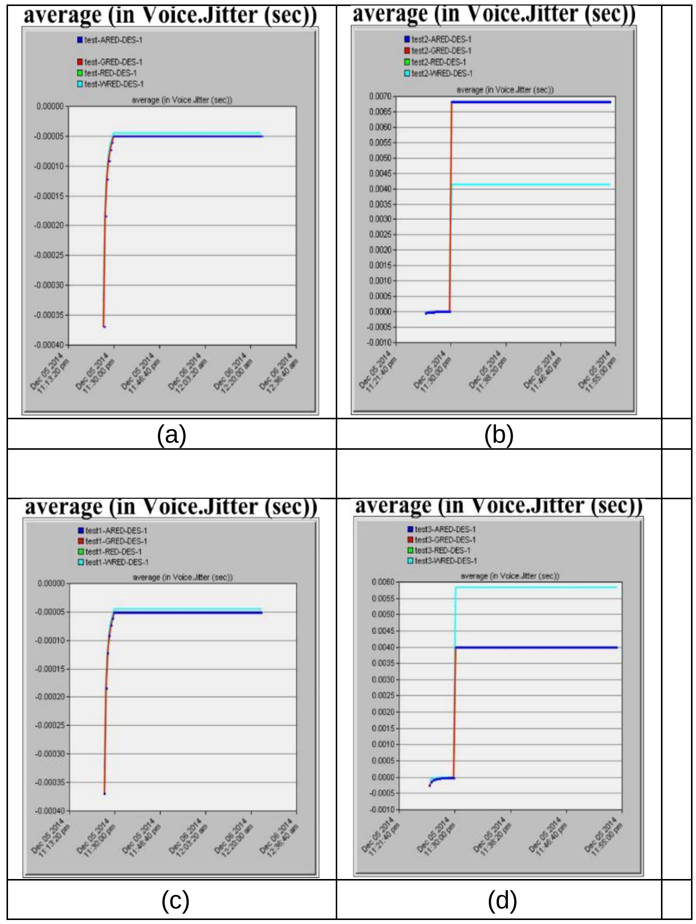
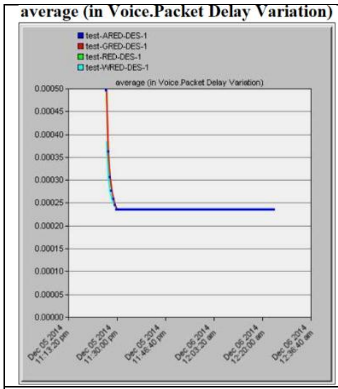
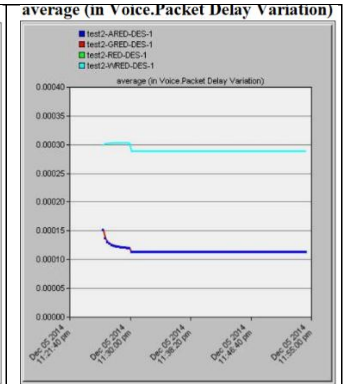


Figure 4.13 *average in voice jitter (sec) (a) G711 codec, (b) G723 codec, (c) G729 codec, (d) GSM codec.*

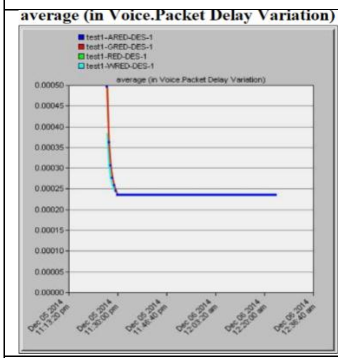
4.1.4.2 Voice Delay results:

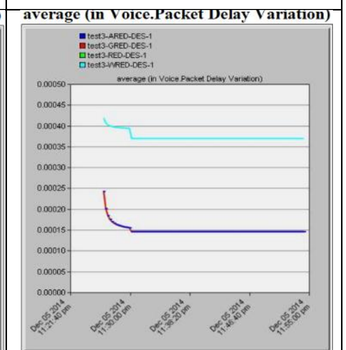
The figure 4.14 samples shows the delay (sec) for the voice packets here we can see in 4.14that almost all the mechanism fall in the same level in the case of part (a), in the case of part (b) we see that worst delay appears when we use WRED but GRED, RED, ARED less delay, in the part (c) we can see that almost all the mechanism fall in the same level, in the part (d) the worst delay appears when we use WRED but ARED, GRED,RED less delay.





(a) (b)



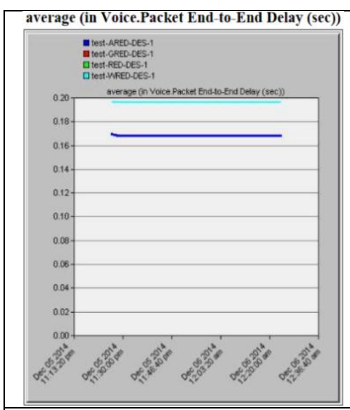


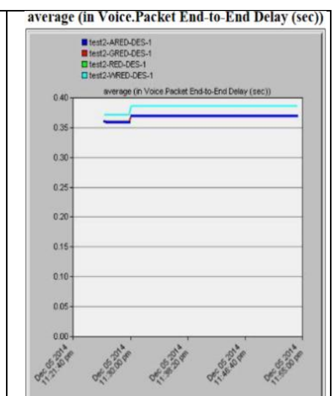
(c) (d)

Figure 4.14 *Average in packet (delay) (sec)) (a) G711 codec, (b) G723 codec, (c) G729 codec, (d) GSM codec*

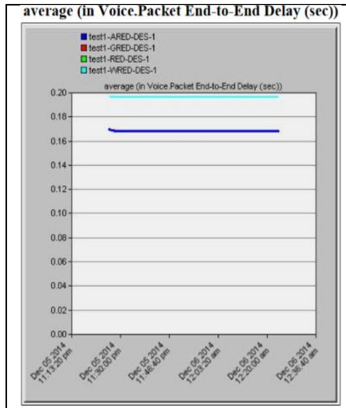
4.1.4.3 Voice End to End Delay results:

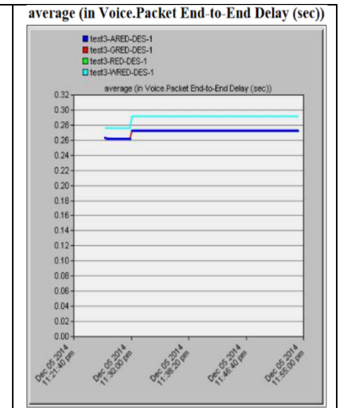
The figure 4.15 samples shows the end to end delay (sec) for the voice packets here in 4.15 we can see that worst delay appears when we use WRED but GRED, RED and ARED less delay in part (a ,b ,c ,d) cases.





(a) (b)

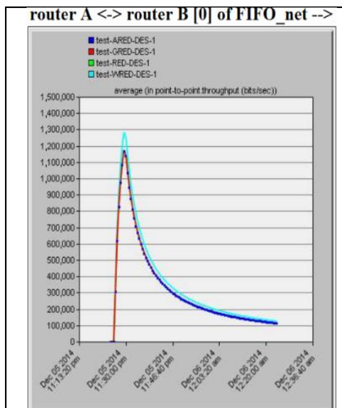


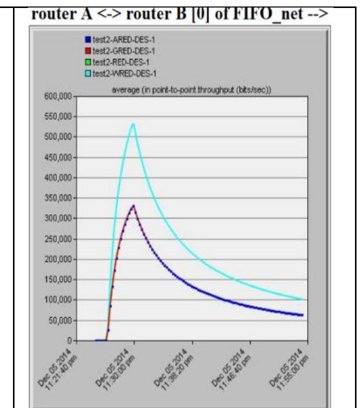


(c) (d)

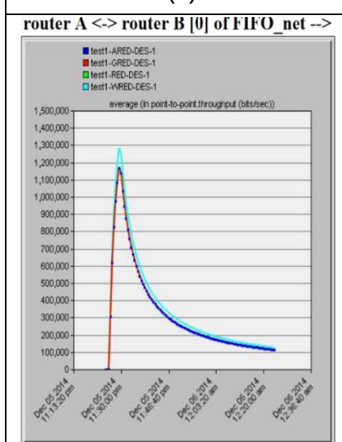
Figure 4.15 *Average in packet (end to end delay) (sec)) (a) G711 codec, (b) G723 codec, (c) G729 codec, (d) GSM codec*

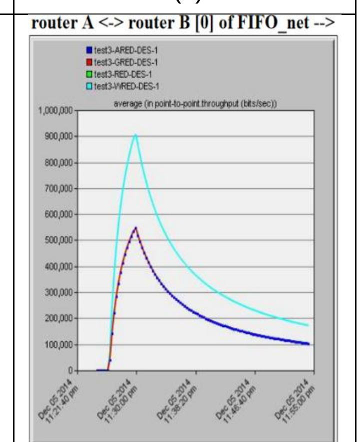
4.1.4.4 Voice throughput user to server results:





(a) (b)





(c) (d)

The figure 4.16 *samples shows the throughput , as we can see in the 4.16 WRED shows best throughput performance in part (a) among the other three parts , which is vary about 1200000 bit/sec in part (a) and part (c), and 550000 bit/sec ,900000 in part (b) and part (d) respectively.*

Figure 4.16 average throughputs for sources and destination (a) G711 codec, (b) G723 codec, (c) G729 codec, (d) GSM codec

5.1 Conclusion:

This study evaluated the performance of VoIP over Ethernet WAN by applying AQM mechanism under various voice codecs scheme in OPNET simulation tool. A comparison has been conducted between four of (voice codec scheme + This comparison aimed to identify which codec offers more satisfactory performance measures for application like VOIP. A decision which codec offers more satisfactory performance measure results is only made depending on varying the number of users, The result shows a selection of G.711 and G729 codec in a simulation gives a significant result for the performance of VoIP that codec G711 and G.729A has acceptable throughput and less deviation of received to transmit packet as compared to GSM and G.723 also average delay like end to end delay and Voice jitter is lesser in codec G711 and G.729 as compared to the other two referenced codecs. And the configuration of ARED, GRED, RED respectively algorithms performs better, achieving a lower discard rate and lower overall delay.

5.2 Recommendations:

This study used the comparison according to wire network component , it recommend that to use some other network component like wireless network because wireless is much worse than wire network, to see how this mechanism will deal with the VoIP packets and codecs scheme while there is another types of network infrastructure .

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