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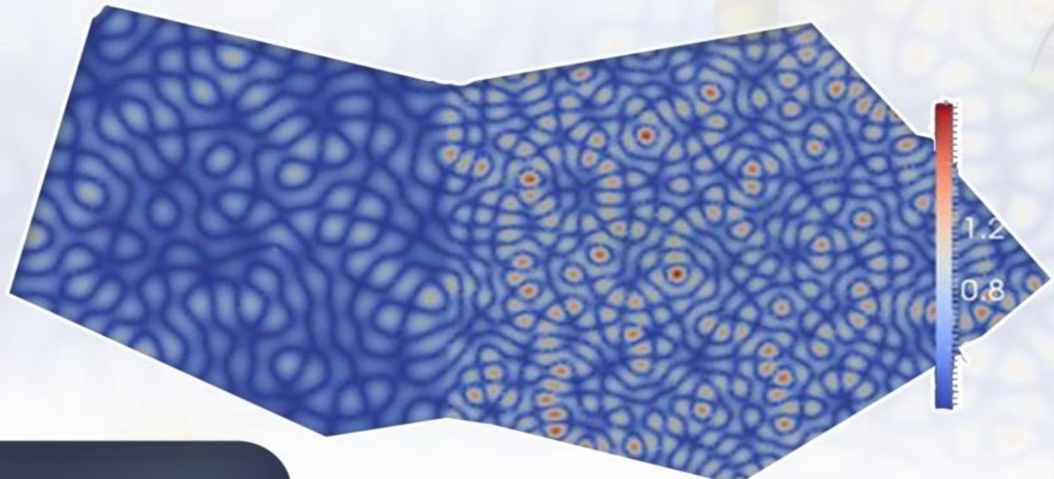

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The Effect of Decreased Voice Quality in a VoIP Network

Jazya M. Ahmad and Aziza A. Omar

Computer Sciences Department, Sirte University, Sirte, Libya.

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A B S T R A C T

Voice over Internet Protocol (VOIP) is a telephone system telecommunication technology that uses an IP (Internet Protocol) network. VoIP offers numerous advantages for customers and communication, including cost savings, phone and service portability, mobility, and integration with other applications. Because voice is sensitive to delay and jitter, bandwidth must be guaranteed while it is being transported. There are some issues with Quality of Service (QoS), the most significant of which is latency in VOIP. The purpose of this paper is to investigate the effect of delay on speech quality in Voice Over Internet Protocol (VOIP). One source of delay is the buffer that already exists in the original multiplexing'-multicast node. There will be no buffering to store packets in the proposed multiplexing'-multicast algorithm, so any packet received will be sent immediately unless the packet arrives after a specific time or in a different order, where OPNET IT Guru Academic Edition simulation was used to analyze network performance.

1 Introduction

Voice over Internet Protocol (VoIP) is a method of transmitting audio and video over a network (Alhayajneh, A. et al.,2018) VOIP allows users to make and receive voice calls over the Internet. IP telephony is the transmission of voice, fax, and other services over packet switched IP-based networks (Odii, N. 2017) The VOIP infrastructure consist of telephone, control node, gateway node and the IP based network , the goal of VOIP is providing voice transmission over network, Reducing the latency is of prime importance to voice data services as it directly affects the acceptance trend of VoIP among mass consumers (Karim, K. 2020) VoIP converts analogue voice signals into digital data packets and supports real-time communication, including packet generation of digital singles using the Transport Control Protocol (TCP), User Datagram Protocol (UDP), and Internet Protocol (IP), as well as packet reception and analogue single reconstruction at the destination. The IP networks used the transmission control protocol (TCP) for transmission of the packet.

1.1 Multiplexing Algorithm

A multiplexer is a device that allows digital information from several sources to be routed onto a single line for transmission to a single destination (Vijayakumar. E et al, 2019) Multiplexes have the advantages of being able to transmit a high number of signals over a single channel, increasing management conversation visibility, and allowing users to utilize the entire bandwidth of the channel. Multiplexer has only one output which is connected to the single communication channel (Singh, C. 2018).

Corresponding author: E-mail: jazamoftah@su.edu.ly

1.2 Multicast Algorithm

Multicast is one of the mechanisms by which the power of the Internet can be further harnessed in an efficient manner (Pati, M. 2020) The benefit of multicast technology is that it supports dispersed applications and allows for greater bandwidth consumption. However, the disadvantages are that the drops are to be expected, there is no congestion avoidance, and data is duplicated, multicasting deals with forwarding packets to a set of destinations (Farhan, K. A et al., 2019).

2 Materials and Methods

The main goal of proposed algorithm is reducing the delay as most as we can. When we use the original algorithm, which combines multiple channels into one, the multiplexer sends all packets at the same time, and if one packet is delayed, all packets will wait for it. The proposed algorithm that will be used is sending the received packet immediately after receiving, will not be buffering to store the packet in it.

2.1 Original Multiplexing Algorithm

The transmission medium transports the signal from the sender to the receiver. To make the best use of that medium, we must ensure that the channel's bandwidth is used to its full potential. Multiplexing is a method of dividing the available bandwidth of a single transmission medium into multiple channels. (Liew, H et al 2002) This algorithm will discuss a number of variables that explain how multiplexing works.

Steps:

- Received: This variable refers to the number of packets received in order.
- Played: This variable indicates the sequence number of the packet being played out.
  - Over-run: This variable informs the play-out buffer that it is full.
  - The Time Division Multiplexer (TDM) which operates asynchronously to the packet arrival processing and is not defined here, empties the play-out buffer.
- Under-run: This variable indicates to that the play-out buffer been Expected.
- D: This variable indicates to the difference between expected and received.
- L: This variable indicates to difference between sequence numbers of packet

Upon receipt of a packet
if received = expected { treat packet as in-order }
if not over-run then
place packet contents into play out buffer
else
discard packet contents
set expected = (received + 1) mod 2^16
else
calculate D = ((expected-received) mod 2^16) - 2^15
if D > 0 then { packets expected, expected+1, ... received-1 are lost }
while not over-run
place filler (all-ones or interpolation) into play out buffer
if not over-run then
place packet contents into play out buffer
else
discard packet contents
set expected = (received + 1) mod 2^16
else { late packet arrived }
declare "received" to be a late packet
do not update "expected"
either
discard packet
calculate L = ((played-received) mod 2^16) - 2^15
if 0 < L <= R then
replace data from packet previously marked as lost
else
discard packet

Figure (1). Original Multiplexing Algorithm (Liew, H et al 2002).

2.2 The Multicast Algorithm

The data in multiplexing divides a communication channel into several number of logical channels where in broadcast, the sender will send a copy to all the member of network, but sometime, there is no need to send to all the members for many reasons, such as, some data is not supposed to reach some employees and other reason, the overload will make an effect on the performance of the network. In multicast transmission, the sender transmits only one copy of data that is delivered to multiple receivers. VoIP's request is based on its capability to make possible for voice and data convergence at an application layer. The big challenge in multicasting is to minimize the amount of network resources utilized to compute and setup multicast trees. When the packets are small and consist of a small amount of data, the loss of packet will have less impact of the quality.

There are many variables that will be mentioned in this algorithm that explain the way that Multicast algorithm works.

Steps:

Socket: through the socket, adding the member to the group or remove the member from the group will be done through the socket.

Port: every data transferred through a network use a port number to distinguish it from other data.

IP address range: in this algorithm, we are going to use IP address included in this range 224.0.0.100.

The Server Side
Open the connection // through the socket
Chose the group that will receive the data
if sender = expected
receive the sender packet
set expected = (received + 1) mod 2^16
else
calculate D = ((expected-received) mod 2^16) - 2^15
Chose the type of socket
Chose the protocol // in our algorithm UDP will be used.
Put the IP multicast range // such as 244.100.0.1 that mentioned before
Put the port number //such as 9050 that mentioned before
IP End Point = new IP Endpoint (range, port)
Data=" this variable will include the data"
Server.send (IPEndPoint, Data)

Figure (2). Multicast Algorithm.

2.3 The Proposed multiplexer Algorithm

The problem can be solved using multiplexer – multicast algorithm that combines the data which comes from several streams into a single large packet, where the packet that comes first is sent over multiplexer for multicast. The technique that will be used is combines the data which comes from several streams into a single large packet, where the packet that comes first is sent over multiplexer for multicast**.** The differences between the two algorithms, the source of latency in the original algorithm is a buffer, and any packet that comes, the algorithm will check the buffer to see whether there is a space in the buffer or not, so, the elements in the buffer will be counted, then the decision will be making. This process will time consuming will result in increasing the latency algorithm, but in the proposed algorithm, there is no such a processing for the packet to be put in the buffer. The Proposed algorithm is show in Figure (3).

Upon receipt of a packet
if received = expected
send the received packet
set expected = (received + 1) mod 2^16
else
calculate D = ((expected-received) mod 2^16) - 2^15
if D > 0 then
{ packets expected, expected+1, ... received-1 are lost
Send the received packet
set expected = (received + 1) mod 2^16}
else { late packet arrived }
discard packet

Figure (3). Proposed Algorithm.

2.4 VoIP Deployment

Flowchart below gives an 8-step methodology for a successful VoIP deployment. The first four steps can be performed in parallel. Before embarking on the simulation in step 7, step 5 must be carried out which requires upfront modification to the existing network. Both step 6 and 7 can be done sequentially. The final step is the pilot deployment.

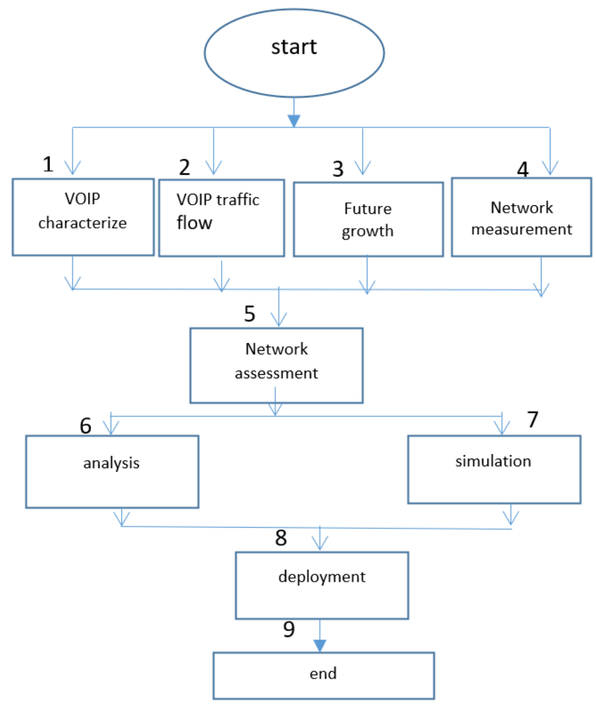

Figure (4). Flowchart Illustrating VoIP Deployment

2.4.1 VoIP Traffic Characteristics, Requirements and Assumptions

The configuration of the workstations will be as the default configuration except that we need to select the supported profile in the workstations in subnet1 which will send the packets, and also, we need to configure the supported services in the workstation in subnet2 in order to receive the packets

2.4.2 VoIP Traffic

Using the predefined voice application as a model for VoIP traffic in OPNET becomes a approach. In OPNET, an application is a collection of tasks, each of which is defined as a set of phases, each of which occurs between two endpoints and has configurable traffic behavior. Applications can be defined and configured using the Application Definition, the application are already defined usually give some flexibility to the user to configure their attributes such as, the configurable parameters.

2.4.3 Growth Capacity

The growth expected of the network has to be all taken into consideration to extrapolate the required growth capacity. To achieve the desired output, The OPNET simulation has been configured to be 20 minutes. Network resources of the router, Workstation, VoIP Application and Profile Settings. This percentage in practice can be variable for each network resource and may depend on the current utilization and the required growth capacity.

2.4.4 Network Measurements

In this step, the existing network based on the existing traffic and the requirements of the new service to be deployed. Immediate modifications to the network may include adding and placing new servers or devices, upgrading PCs, and utilized links.

2.4.5 Network Assessment /Modification

We need to change the default configurations of some network components in order to develop a realistic model. The following Figure (5) describes the configuration changes applied to each of the components of the network.

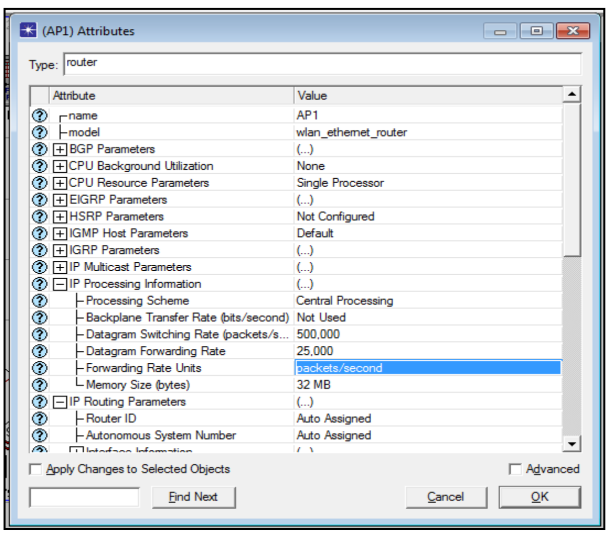

Figure (5). The configuration changes to components the network

2.4.6 Analysis

In order to gain accuracy of the simulation results, five simulation replications were run. Simulation replications were sufficient. Each simulation replication produced very similar graphical results.

Figure (7 and 8) show that the proposed algorithm has a shorter end-to-end delay than the original algorithm. The original algorithm's source of delay is the buffer that already exists in the multiplexing node, as well as the process that can be done in the packets before sending them to the multicast node.

Bandwidth Bottleneck Analysis

A bandwidth constraint is a phenomenon in which the performance of a network is limited because not enough bandwidth is available to ensure that all data packets in the network reach their destination. A network bottleneck occurs when a system working on a network delivers a higher volume of data than what is supported by the network's existing capacity.

Delay Analysis

This delay limits the amount of time that can be sustained. It should be noted that the primary goal is to determine the VoIP network capacity, that is, the maximum number of calls that the existing network can support while maintaining VoIP QoS. This can be accomplished by gradually adding calls to the network while monitoring on the VoIP delay threshold or bound. When the end-to-end delay, including network delay, exceeds 150ms, the maximum number of calls can be determined.

Comparison between Original and proposed algorithm

For the proposed algorithm, the latency will be reduced since the received packet will be transmitted directly to the next receiver as we will clarify it in Table (1).

Source	Original Algorithm	Proposed Algorithm
Source of Latency	Existing in the following: If received = expected If not over-run then Place packet contents into play out buffer - There is a buffer to be checked for a space and the elements in the buffer will be counted. - Consuming Time. -Increase Latency, packet stay in buffer for a *T* time.	If received = expected Send the received packet - No process to keep packet in buffer. - There is no buffer

Table (1). A comparison between Original and proposed algorithm

The source of delay in the original algorithm existing in the following:

if received = expected

if not over-run then
    insert packet contents into playout buffer

there is a buffer, and any packet that comes, the algorithm will check the buffer to see whether there is a space in the buffer or not, this process will consume a time which as result will increase the latency.

In the proposed algorithm, there is no such a processing for the packet

if received = expected
send the received packet

2.4.7 Simulation

Models were simulated using OPNET on a Toshiba laptop running Windows 7.
The simulation approach uses the popular OPNET IT Guru Academic Edition simulation. There are many features of OPNET such as comprehensive library of network protocols and models, GU I interface, graphical results and statistics.

Modeling the Network

The popular OPNET IT Guru Academic Edition simulation, Release: 9.1, is used in the simulation approach. OPNET Modeler already has a large number of models of available system components, as well as various network simulator configuration capabilities. As a result, the simulation of a real-world network environment is very close to reality. OPNET also includes a comprehensive library of network protocols and models, a GU I interface, graphical results and statistics, source code for all models, and many other features. The fact that OPNET is offered free of charge to academic Universities and institutions is even more significant for its popularity in academia. This is why OPNET has an academic advantage over DES NS2.

Simulation Procedure

Subnet1 has been modeled as a subnet that encloses a router and five Ethernet workstation used to model the LAN users activities. AS it shown in the subnet 1, the five workstations will generate the traffic, that means the workstations that labeled with SUB1_WS1, SUB1_WS2, SUB1_WS3, SUB1_WS4, and SUB1_WS5 is a source for sending VoIP calls. AS it shown in the subnet 2, the five workstations will received the traffic, that means the workstations that labeled with SUB2_WS1, SUB2_WS2, SUB2_WS3, SUB2_WS4, and SUB2_WS5 is a sink that will receive VoIP calls**.** The simulation run can take up to 20 minutes. The simulation approach, as it will be demonstrated, is automated.

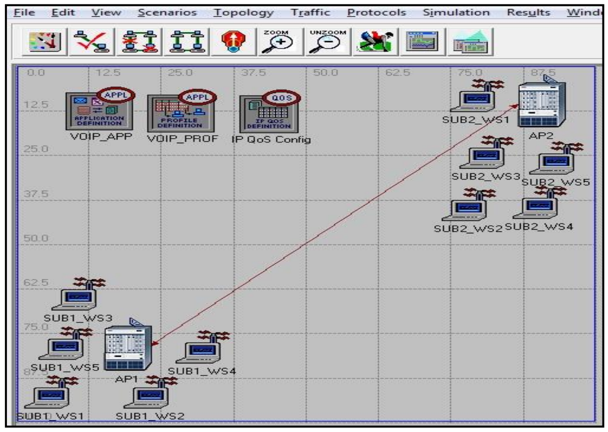

Figure (6). Shows the described topology

2.4.8 Pilot Deployment

A pilot deployment is the place for the network engineers, support and maintenance team to get firsthand experience with VoIP systems and their behavior. During the pilot deployment, the new VoIP devices and equipment are evaluated, configured, tuned, tested, managed and monitored, During the pilot deployment, the new VoIP devices and equipment are evaluated, configured, tuned, tested, managed and monitored.

3 Simulation Results

Using the predefined voice application as a model for VoIP traffic in OPNET becomes an approach. In OPNET, an application is a collection of tasks, each of which is defined as a set of phases, occurs between two endpoints and has configurable traffic behavior.
The graph (7) appears that the maximum point of the delay after running the simulation for 20 minutes is approximately 1.65 second in the original algorithm.

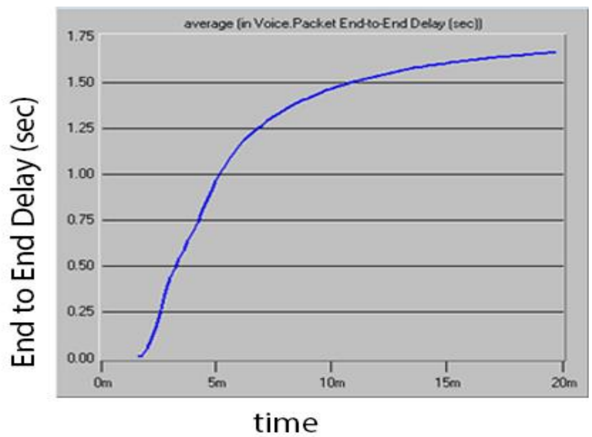

Figure (7). End-to-End delay in the original algorithm

Figure (8) shows how the End – to – End delay increases as the number of calls increases over the network's lifetime. The maximum point of the End to End time is approximately 0.5 second. The processing that will be done in the multiplexer node is the cause of this minor delay.

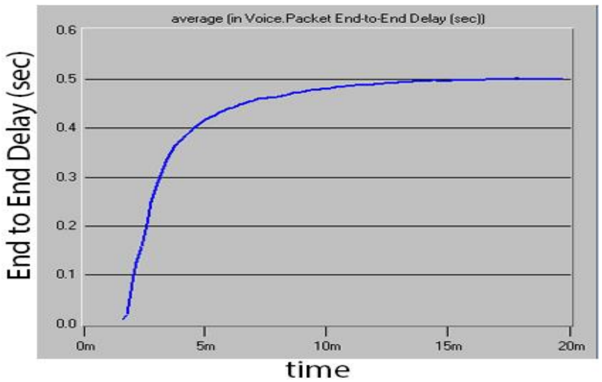

Figure (8). End-to-End delay in proposed algorithm.

4 Conclusions

The simulation route takes about 20 minutes and produced 46975482 events. According to Figure (7 and 8), the proposed algorithm has a shorter end-to-end delay than the original algorithm. The buffer that already exists in the multiplexing-multicast node is the source of the delay in the original algorithm.

Conflict of Interest: The authors declare that there are no conflicts of interest.

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