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Evaluate How to Minimize the Effects of LTE Network Congestion for VOIP

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Abstract: *Long-term evolution[1] is the fourth generation technology of cellular networks in which the transfer rate of voice, data and video is very high. VoIP[2] enables people to have phone conversations online. This article aims to find ways for high quality service for users, especially when the network is busy, since VoIP is a real-time service and competing over access to communication channels leads to packet or failure in packets. Hence, in cases when these parameters are higher than toleration level, the service is influenced and network congestion occurs. So, as we can infer, traffic control and service quality are connected and proportionate and an improvement in one of them improves the other. Researchers try to improve service quality parameters to control congestion. This paper focuses on end-to-end packet of voice on the internet protocol in centralized LTE networks. In order to evaluate the packet of E2E[3], simulation is done using NS3 tool. Three different scenarios are defined; one is network-based, one includes VoIP and the third includes VoIP with FTP[4]. E2E packet and packet loss for every scenario has been measured. A comparative performance analysis has been simulated for both networks using output graphs; and VoIP performance in LTE has been discussed.*

Keywords: *Long-Term Evolution Network, Voice, The Internet Protocol.*

INTRODUCTION

The method for telephone communication is changing. Today, for distant telephone connections, VoIP technology is used. VoIP is method to change analogue signals into digital ones via the internet. VoIP has the capability to change the telephone system operation methods completely. There are numerous companies that offer VoIP and their number is increasing. At the moment cellular networks that offer mobile services are expanding very fast. Right now in developed and developing countries mobile networks based on the third generation and higher technologies like HSPA and LTE are made operational and they offer services to clients (Moyano et al., 2015). They provide high-speed information transfer. The competition between these mobile networks has made them offer various services in the market. The idea of future generation of networks with the intention of providing multi-media at any time-anywhere has created a packet-based technology setting with complete mobility in communication networks that is pursued by network and service providers. Thus, the next objective of communication networks is 4G generation of mobile networks called LTE-Advanced

[1] LTE
[2] Voice over Internet Protocol
[3] End to End
[4] File transfer protocol

networks. In this new network bit rates as high as 1Gbps with complete mobility is intended (Baynat and Nya, 2016).

The intention for providing the models in this paper is discussion over traffic-related issues for VoIP alone and alongside other traffics in the LTE network. E2E packet for VoIP is a critical issue for real time applications on the internet. Real-time applications require more resource (Chen, Lin and Lin, 2011). This paper discusses technical and required problems for VoIP in computer networks. It also argues the challenges about having to face computer networks for the transfer of VoIP applications. It provides a descriptive idea about VoIP on LTE and the parameters of design and application on LTE. In this paper, quantitative and qualitative analyses about packet in LTE is performed in a simple and understandable model for those who intend to investigate more on the matter (Ko et al., 2016; Casey et al., 2013).

- **VoIP introduction**

VoIP is a technology that provides vocal communications on computer networks like the internet and any other IP based networks. VoIP technology offers to decrease telephone costs using internet switching packets.

In VoIP engineering, the main audio signal is sampled and encoded in the end of sending process into a steady bitrate digital flow. This flow of compressed digital data is divided into equal packets for easy playback. Every packet contains audio data, departure address information and predicting destination address. The packet flow is recreated by means of a time-tag. These packets are then put on a digital to analogue packet switch to be understandable by human ear (Brahmbhatt, Mann and Rawat, 2017).

- **Performance evaluation**

In real world scenarios, the evaluation of network performance is very important. To investigate the real scenario, different simulators are used. NS3 is a commercial simulator. We used NS3-3.26 version of it in this article because of its efficiency and reliability (Kim, Niyaz and Javaid, 2014).

Configuration of network model

*** Network components**

By network components we mean network models that are operationable on NS3.

*** Network congestion creation**

In order to create an application in NS3 called application definition attribute. This includes default applications that can be personalized according to the users' needs. There are numerous default applications like HTTP, email, video, FTP, voice, database, etc. There are two applications (VoIP and FTP) that are defined according to applications' properties. FTP application is modeled to simulate background traffic. Since this application transfers file at specified intervals, constant 60 is regulated to created FTP traffic (Luo, Liu and Xie, 2009). Voice application is designed with configuration table. VoIP application uses G.711 encoding and interactive sound to create VoIP calls.

Table 1: FTP application parameters

Amount	Properties
50%	Order combination
Constant 60	Enter request time
1000000000 constant	File size (bite)
FTP server	Server sign
Best effort	Service type
Does not have	RSVP parameters
Not used	Default applications

- **Simulation general parameters**

Table 2 shows general LTE parameters used in all simulation models. Another important element of configuration is mobility which is used for work unit mobility models (Stafecka, Lizunovs and Bobrovs, 2018; Teković, Pešut and Morić, 2013).

Table 2: LTE parameter

Amount	Parameter
1 (GBR)	Qos class tag (audio)
6 (non-GBR)	Qos class tag (FTP)
1 Mbps	Guaranteed sending bitrate (bps)
1 Mbps	Guaranteed receiving bitrate (bps)
1 Mbps	Maximum sending bitrate (bps)
1 Mbps	Maximum receiving bitrate (bps)
1920 MHz	Base sending frequency (GHz)
20 MHz	Sending bandwidth (MHz)
7 Symbol on slot	Sending periodic prefix type
2110 MHz	Base receiving frequency (GHz)
20 MHz	Receiving bandwidth
7 Symbol on slot	Receiving periodic prefix type

- **Simulation scenarios**

Three simulation scenarios are created as follows. The same topology of the network is considered for different simulation scenarios.

1. **Scenario A: Base VoIP network**

Base VoIP networks are designed in NS3 simulator using different network elements. The radius of the cell here is 1 km. Network elements used during network model designing include simulation, LTE configuration, application configuration, profile configuration and mobility configuration. Other network elements include eNodeB, evolved core pack and network stations. There are four simulations based on the mobility of the knots that include:

Table 3: Simulation items of base VoIP network

Item	Bandwidth (MHz)	VoIP traffic load	Cell radius	Speed (m/s)
1	20	50	1	Constant 0
2	20	50	1	10
3	20	50	1	20
4	20	50	1	50

2. **Scenario B: Compressed VoIP network**

In compressed VoIP network 95% of the traffic in this scenario is audio. The aim for this scenario is investigating the effect on audio when traffic is about 95%. VoIP conversations are created for 95% traffic in order to obtain end to end, sending and receiving traffic. Table 4 shows simulated compressed amounts of VoIP that is based on mobility of work stations.

Table 4: simulated compressed amounts of VoIP

Item	Bandwidth (MHz)	VoIP traffic load	Cell radius	Speed (m/s)
1	20	95	1	Constant 0
2	20	95	1	10
3	20	95	1	20
4	20	95	1	50

3. **Scenario C: Compressed VoIP with FTP network**

Combined traffic (VoIP and FTP) is created in this scenario in which FTP is modeled to introduce background traffic. The guaranteed bitrate (BGR) is used for audio and non- guaranteed bitrate is used for FTP traffic in the simulation scenario. It is worth mentioning that FTP workstations are stable in all simulation cases,

Table 5 shows compressed VoIP simulation with FTP network. Two criteria, being mobility workstations and different traffic loads are considered.

Table 5: compressed VoIP simulation with FTP network

Item	Bandwidth (MHz)	VoIP traffic load	Cell radius	Speed (m/s)
1	20	80	Constant 1000000000	Constant
2	20	80	Constant 1000000000	20
3	20	80	Constant 2000000000	Constant
4	20	80	Constant 2000000000	20

- **Simulation time**

Simulation time for all cases was 600 seconds and all voice applications were performed for 100 seconds of the simulation time. All calls were started simultaneously.

4. Network operation analysis in different scenarios

4.1. end to end operation analysis

End to end packet means the time required for a packet to go from a node to another. E2E unit is seconds. Generally, three types of packets occur. First, there is the sender's packet; the time required for the packet to be sent from the sender. The other two are network packet and receiver packet. For VoIP applications the packet should not be greater than 150 ms in order to be able to evaluate that the VoIP call is acceptable regarding quality. In this part the result of E2E packets are shown for different scenarios. Scenarios A, B and C are related to base VoIP, compressed VoIP and Compressed VoIP with FTP respectively. In the scenarios A and B, the audio application is defined between the source and the destination. The voice application and FTP are designed in scenario C. All scenarios, resources and destinations start at 100 seconds and profile and configuration times are set on 40 and 60 seconds respectively. In all presented figures below, x axis shows simulation time per second. Axis Y shows end to end packet time.

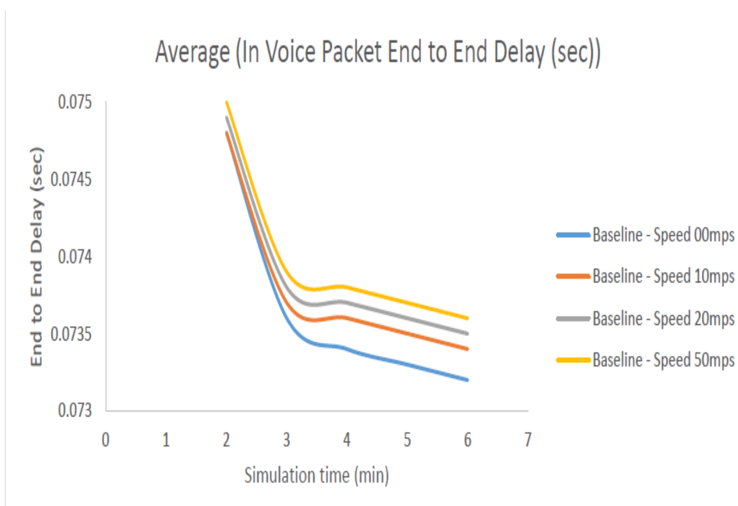

Figure 1. End to end packet of base network

Figure two compares E2E packet in different mobility scenarios in compressed VoIP networks

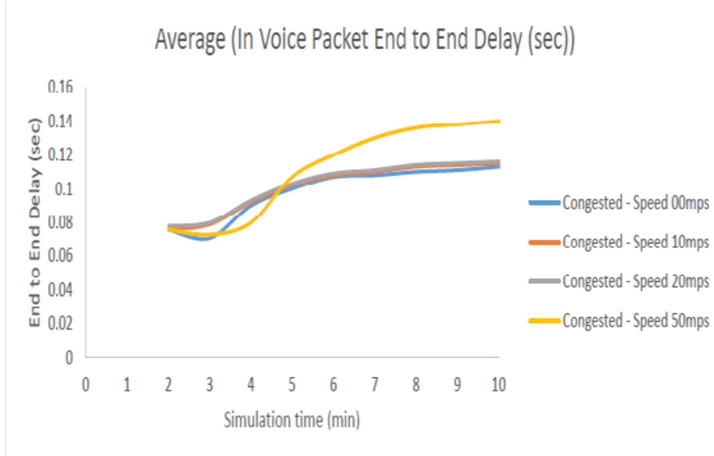

Figure 2. End to end packet of compressed VoIP network

Figure 3 compares E2E packet for different scenarios of VoIP with FTP network.

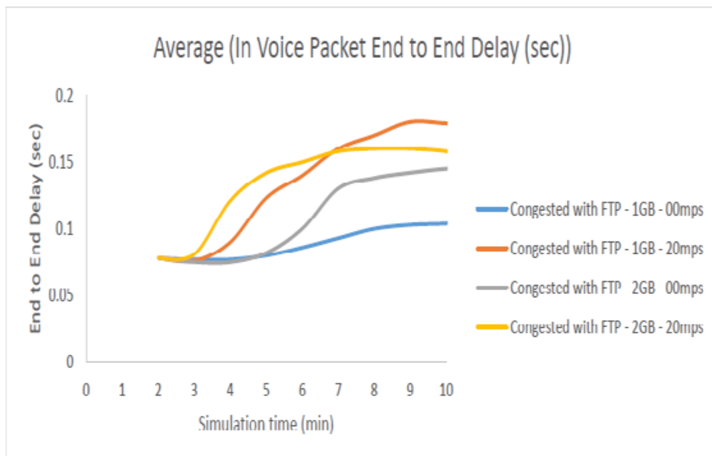

Figure 3. Packet for VoIP with FTP network

- **A summary of E2E packet operation**

In figure 4, x presents four different items and Y axis shows end to end packet in milliseconds. From figure four and observation we can conclude that:

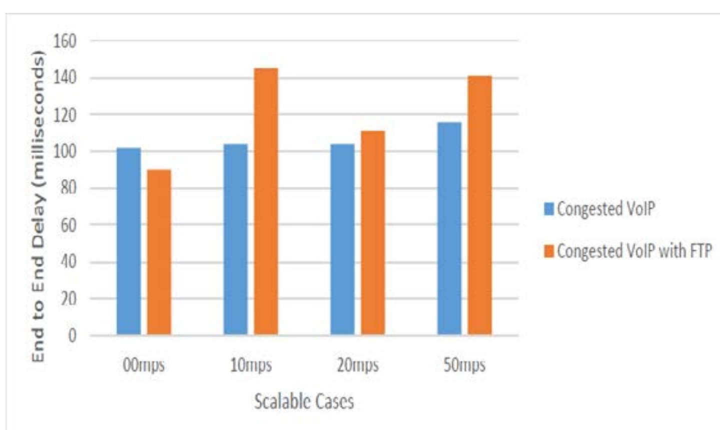

Figure 4: Medium E2E packets between two different scenarios.

For item 1, in general packet for compressed VoIP is 10.5% more than compressed VoIP with FTP. In scenarios 2, 3 and 4 E2E packet in VoIP with FTP is 28%, 8.5% and 17.5% more than compressed VoIP respectively. So, compressed VoIP has a better performance than VoIP with FTP regarding E2E packet in moving and stable knots. Compressed VoIP with FTP acts well regarding E2E packet.

4.2. Packet loss operation analysis

In this part, different ratios of packet loss are presented. In all the presented figures, x axis presents time per second, and y axis presents send and receive traffic in different scenarios of VoIP network.

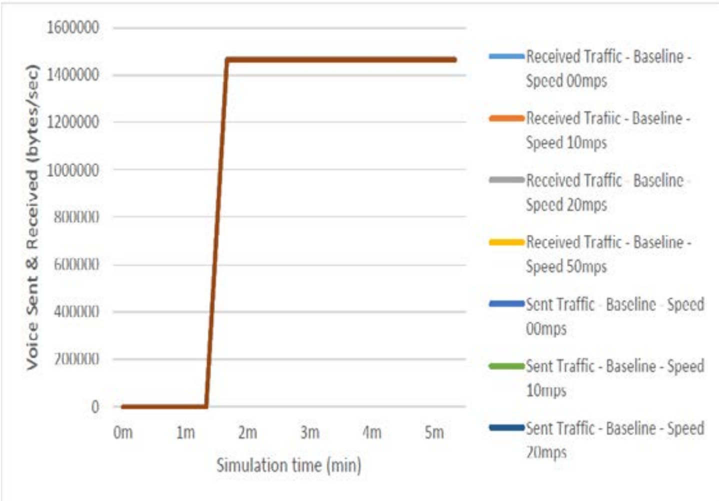

Figure 5. Send and receive traffic from base VoIP network

Figure 6 shows different send and receive traffic scenarios in compressed VoIP network.

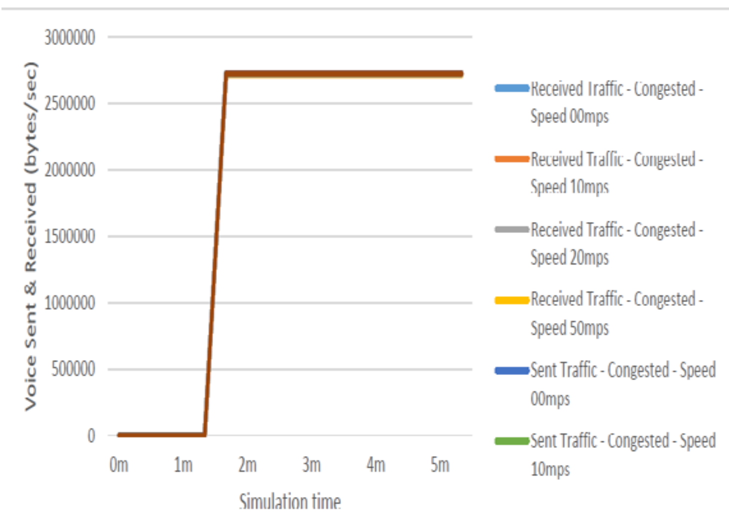

Figure 6. Send and receive traffic of voice in compressed VoIP network

Figure 7, shows different send and receive scenarios for compressed VoIP with FTP.

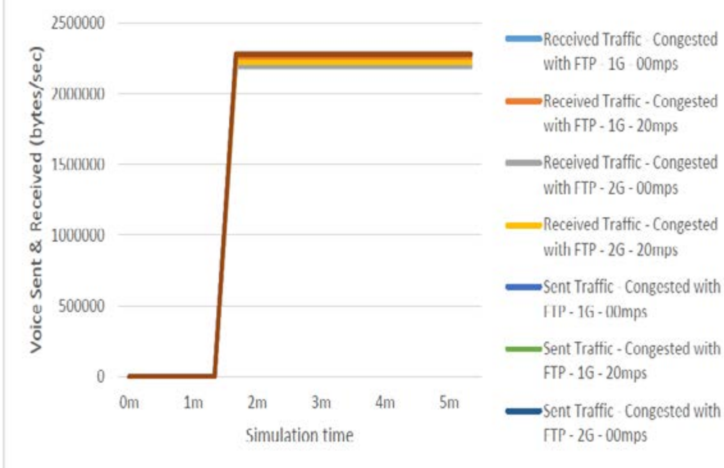

Figure 7. Send and receive traffic of voice in compressed VoIP network with FTP

- **A summary of packet loss**

In this part, packet loss for different compressed VoIP scenarios and compressed VoIP scenarios with FTP are evaluated. In figure 8, axis X shows four different items and Y axis shows packet loss.

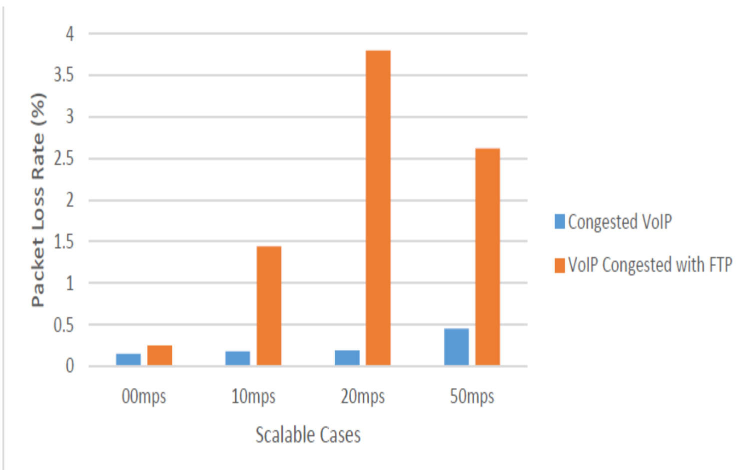

Figure 8: The average packet loss for two different scenarios

Conclusion

In this article end to end packet for VoIP in LTE networks was analyzed. The evaluation was performed in NS3 simulation according to end to end packet and passability. Three network scenarios were simulated: base VoIP network, compressed VoIP network and compressed VoIP network with FTP. As performing the simulation we found out that maximum passability increased with bandwidth increase. Other than the 6 created scenarios, the one with the greatest bandwidth (20 MHz) had the highest passability. After that, quality of VoIP was tested according to E2E packet for VoIP and VoIP with FTP. Four scenarios were created for these evaluations; one with stable node and three with mobile knots (node speed is increasing).

Simulation was improved by NS3 designer[5]. Simulation results show that when the node is not moving, E2E packet for compressed VoIP is a little higher. In other cases, better E2E was achieved because of moving node presence. Packet loss for VoIP network was the minimum regardless of speed. For VoIP with FTP, packet loss for stable node is very minute. Rate results begin with mobility.

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[5] Simulation Instructions, "https://www.nsnam.org/docs/tutorial/html"

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