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Comperative Analysis of Voice Codecs for LTE Communication Networks

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ABSTRACT
4G mobile connectivity protocol, LTE (Long Term Evolution), is lower dormancy and higher data rates. LTE overseers may utilize the current circuit to control voice through the fallback procedure Networks exchanged as LTE is totally Internet Protocol (IP) advancement. Voice over IP administrations (VoIP) are needed to give the voice over IP associations, similar to the Internet, to prevent this. Voice Codec is one of the basic sections of VoIP that can be communicated on Internet and change simple signs into compacted computerized parcels. In any case, there are various sorts of codecs that lift voice streams dependent on transfer speed and application necessities. There is, still, no default codec reasonable to everybody. Some codecs have better effectiveness, with different conditions and therefore the transmission capacity is lower. Every one of these qualities will influence the Ultimate corporate assortment of codecs. In this paper, investigation on various kinds of codecs is held that can be utilized with ns-3 test system in VoIP over LTE. Moreover, we have proposed some framework for examination of VoIP traffic over LTE nature of administration (QOS) by arranging various VoIP codecs. Furthermore, as far as deferral, jitter, parcel misfortune and error-proposed scenarios, the study of the different VoIP codecs across the LTE network has been evaluated.

KEYWORDS
analysis of voice codecs, QOS in VoIP, LTE networks, Internet Protocols

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INTRODUCTION

The green public procurement identifies LTE the 3GPP-engineers have called high Long-Term Evolution as the next level in the development process for Global System Mobile (GSM), a 2G technology, and the 3G-standard Universal Mobile Telecommunications System (UMTS) which supports GSM4 (3GPP) is a fairly modular broadcasting interface. LTE has greatly advanced data rates, reduced latency, scalable bandwidth and compatible cooperation with new generations of GSM and UMTS. The original LTE version has a higher rate of 300 Mb/s. A latent radio network is less than five m/s. The capacity of a radio network increases dramatically compared to earlier mobile networks.

While implementing VoIP for achieving maximum QoS and capacity the adoption of an appropriate voice codec is critical [1]. The quality and performance of VoIP conversation made over a (MANET) registered with SIP proxy which is wireless nodes are accessed [2][3][4][5][6]. Obtaining and sustaining good sound quality is one of the most difficult aspects of VoIP adoption.

A VoIP network's user perceived voice quality must be comparable to that of traditional circuit-based networks[7][8][9][10][11]. LTE provides a broad range of network bandwidths and a wide range of different band provisions in addition to the Time and Frequency Division

Duplex. The number of People who interacted with SIP accounts at the same time in, if the theoretical bandwidth values of voice codecs were used VoIP would be enormous [12]. LTE is also a significant step forward in worldwide telecommunications. In the primary version of the LTE, the greater part of the capacities toward the beginning are customarily considered for fourth generation bearing frameworks. The start of developments in the modern technology and multimedia services in mobile phones has become highly important to consumers. In those days, new mobile network generations have not given the users better broadband networks. In previous generations, voice quality was not higher. LTE provides consumers all of these features at affordable cost with its high data rated ability and increased demand.

One of the most important components of VoIP is the Voice Codec, which converts analogue into digital signals, which may then be delivered on the internet [13]. The VoIP review will classify the gaps in the efficiency of the LTE networks. LTE refers to a significantly higher collection of VoIP clients. The voice quality data transmission in IP packet networks is examined. It examines ways for determining the quality of packet telephony data transfer. A packet model was created and validated to account for VoIP architecture that supports several audio codecs for voice compression [14]. In LTE networks, the integration of packet technologies, codecs and QoS support mechanisms ensures that voice messages are perceived at a level of at least 4 on the MOS scale [15]. In order to provide a decent QoS , QoS treatment is required (QoE). For end-to-end service, LTE provides a different and integrated QoS aware method[16] . The proposed scheme's performance on voice traffic is assessed using the E-Model to assess perceived QoS [17]. The total time taken to pass for a packet is End to End Delay [15].

As the revenue source is foreseen, VoIP service providers are attracted. Present voice communication service providers will compete toward cheaper call rates. VoIP can also manage voice and data over a single network from a technology perspective. Lower calls than are stronger. Some VoIP calls are differentiated from the networks by delivering a fixed rate of output which means that produces a steady output stream unaware of the state of the network, requiring additional channel bandwidth, creating congestion that results[19]. The most popular following VoIP according to the suggestions of the ITU-T G-series is good assessment [20]. For achieving good results and compare to past year researches model is proposed. This delay involves many sources such as delayed routing, delayed propagation, and delayed queuing. The shift in which is the packages. There's no guarantee on the wireless network that all packets can follow path the network leads to packets to reach the destination in various delays out of order[19].

Moreover, there would be no jitter a continuous for instance transmitted on the lost in the network by up to 15 packets due to errors or congestion that 2.5percent loss is reasonable to VoIP. Recently, over all 3G networks, the reliability of VoIP increased. However, the criteria for higher speed data already are fulfilled. The demand for high-speed mobile devices is increasing. The LTE and LTE project developed by Third Generation Partnership is ten times as fast as the 3G network currently in operation for downlink and uplink respectively. The LTE Network intended for the LTE-interface turned on packets the conventional CS (2G or 3G) domain. VoIP is the fundamental communication that is well implemented in circuit-shifted networks. Due to LTE network development, voice service is only transferred to packet change in the network.

In order to provide the voice across the LTE networks it was suggested such as CS fallback, dual transceivers for radio and VoIP end users. Double radio gadget is a procedure where it contains two handsets for voice correspondence and information correspondence. Mobile network can hit and access 2G or 3G core networks offers both CS domain voice and data service.

LTE at the same time, but it has not been the issue standardized and used by various network and industries. This method reveals the operators lack of interest. Then, like a network began using a switched circuit short term solution fallback technique (CSFB) for the provision of data services. The services are provided via SMS Switched networks users with circuit. This is because the switching networks have to be made possible and it affects further delay in setup to provide voice service Voice call, loss of efficiency and also affects use present machinery.

In addition to it as the revenue source is foreseen, VoIP service providers are attracted. Present voice communication service providers will compete toward cheaper call rates. VoIP can also manage voice and data over a single network from a technology perspective. The voice quality is reduced and not delivered on time when the network link becomes poor [21].

The most critical part of voice over Internet Protocols is supremacy. The important evaluation is in terms of lower latency and appropriate throughput for monitoring the efficiency of LTE networks. The VoIP accuracy of 4G LTE was analyzed and discussed. Moreover, QoS parameters include results, latency and jitter are discussed. The output tests how many data in a given time are transferred. Latency specifies how long a data packet has to be transmitted from multiple hops. Jitter is a packet latency variable.

This article analyzes these parameters by discussing various scenarios. VoIP QOS based on a range of factors such as bandwidth, broadband connectivity, codecs (compression) and so there is a possibility that voice quality is low. Moreover, support by full bandwidth and managing elements of network efficiency (delay, error rate, throughput and so on). The main challenge is to ensure high quality voice calling in the transmission of VoIP traffic through the IP network. Addition to it, it focuses, on the consistency calculation for VoIP in terms of jitter, throughput, latency and packet loss of service parameters. Furthermore, VoIP solutions for seamless networking are implemented voice codec transmission of through the circuit device packet.

The next sections is formulated as Related work which describes all the work which is similar or have some drawbacks. The next section is about Materials and Methods in which LTE architecture and codec implementation complete setup is discussed. Moreover, in Experiments and Results section of the four different codecs is evaluated. Lastly, conclution is described in the last section.

RELATED WORK:

Two simulators are discussed in the related work. One is Simulator for LTE-EPC network and the other one is network simulator 3.

a) Network simulator 3:

The most widely used tool for evaluation of the various network terminologies where the construction in the actual world is expensive and time-consuming and used to determine new research theories and hypotheses. Network simulators such as are available. The computing time used by ns-3 relative to other simulators is lower and the fastest simulator has been demonstrated. However, the cost of using ns-3 simulators is higher as well as the complexity of simulators as mistakes in programming are the major issues [23][24][25][26].

The scalable architecture of ns-3 and the core of the ns-3 have multiple practical simulation model systems in the present text. The center is LTE, Wi-Fi and WiMAX support. It's an open source network simulator. It is fully written and is not required in C++ and Python languages, Scripting language separate to ns-3.The newest version of ns-3 is the more practical, well-organized ns-3.27 which supports parallel simulation and has improved feature setting. In OFDMA typical schedulers are used are round robin and the proportional fair.

Moreover, in ns-3 environment it uses the 5G-NR LENA module with various scheduling mechanisms to evaluate downlink access. However, these schedulers do not work on short term fairness [27]. On/off software is an application that has been used to send VoIP traffic via LTE to different codecs. It is an on/off pattern traffic generator.

In order to identify the period off and state, it has been seen that during "off" and constant state no traffic is created. Continuously delivering flow, which may be described by the "package size", "data rate" offered, the LTE network, using the preset packet size and data rate, for codecs. This analysis focuses on the purpose of evaluating the performance of various voice codecs, using the simulator NS3 for different options. The construction in the actual world is expensive and time-consuming and used to determine new research theories and hypotheses.

b) Simulator for LTE-EPC network (LENA):

LENA offers many applications such as mobility control, uplink and downlink schedulers architecture and output assessment, load equilibriums, radio algorithms. However, for better signals fresh technologies are needed which are not available [28]. It consisting mainly of 3 EU is a UICC (SIM) mobile device which manages communication and terminal functions in order to end data streams.

Imitating LTE networks is the most effective way to provide authorities the information the information they need to make smart conclusions. However, existing smart phones don't have this functionality [29]. The Radio Contact between EPC and EPC is managed by E-UTRAN (Evolved-Universal Radio Access network). When compared to the system of universal mobile telecommunication.

LTE has simpler design (UMTS). EUTRAN is made up of many evolved NodeBs (eNBs) which are the LTE radio network's base stations. However, the UMTS are not broadband and they are way expensive and therefore the issues are not resolved [30]. Any codec has been delayed at the destination node.

MME uses signaling messages to monitor high-level operations such as setup of the bearer, mobile protection, etc. It only transmits for any packet flow in the LTE network. LTE simulator (open source) is made to simulate a virtual LTE network, to connect genuine hosts via a real wired link. However, it only transmits flow of the packets in LTE network [31].

MATERIAL AND METHODS:

In the environment of LTE architecture different voice codecs have been examined in MATLAB. Moreover, complete simulation process setup is described.

In Figure 1 as shown the E-UTRAN comprised for Evolved-UMTS Terrestrial Radio Access Network which has the only one component that is known as the base stations or the eNB (eNodeB). It also manages the radio communication between the evolved packet core and the mobile. From UMTS and GSM, the HSS which stands for Home Subscriber Server component has brought forward. This is the main or the central database that have the all network operator's subscriber's data and information. Using the interface SGi, With the outside sector the Packet Data Network is interacted.

Network packet is each found by an Access Point Name. The gateway PDN performs exactly similar functions as support node of GPRS (SGSN) and (GGSN) with UMTS, GSM. The (S-GW), which stands for Serving Gateway, is responsible for forwarding information or data between the PDN gateway and the base station, and so functions as a router.

Furthermore, the MME (Mobility Management Entity) governs the mobile's high-level functioning through signaling messages and the Human Subscriber Server. The PCRF, is depicted in the diagram and is used for making policy control decisions and guiding flow-based charging features. in the PCEF which stands for (Policy Control Enforcement Function) and which is located in the PP-GW. S5/S8 are the two different implementations that is used for the PDN gateways and the serving interfaces. S5 and S8 are two different implementations, with S8 being used when the two devices are on separate networks and S5 being used when they are on the same network.

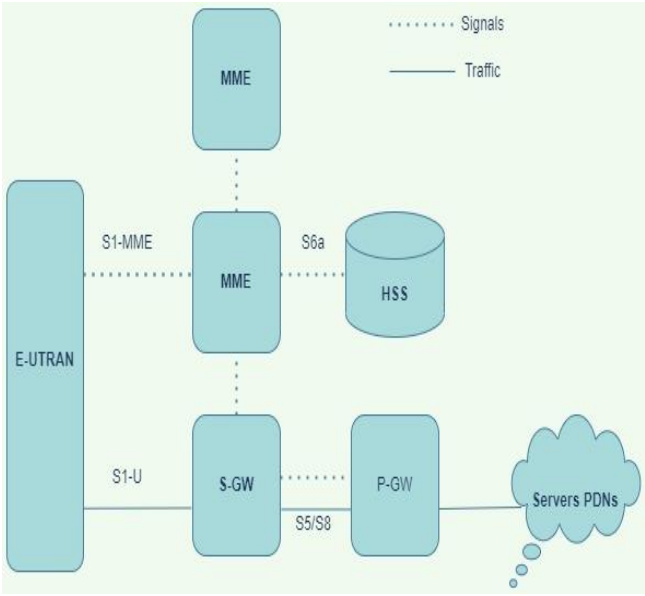

Figure 1 LTE Architecture

For choosing the best codec for Voice over Internet Protocol (VOIP) Conversation the four different codecs that is G.711, G722, G723 and G729 is have been evaluated. Analyzing all these codecs one by one, under the circumstances, the best one which uses minimum bandwidth and gives maximum frame size is chosen and used for VOIP Conversation as shown in the figure 2.

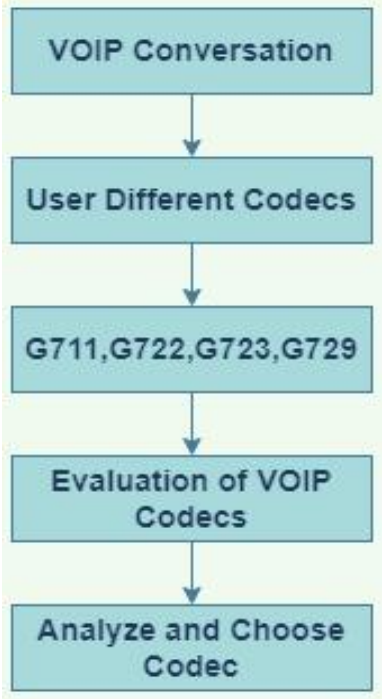

Figure 2 Criteria of codecs for Implementation on LTE networks

As shown in the figure 3 LTE architecture the LTE model is assisted by ns-3 and the LENA team offers easy LTE-EPC for Ns-3 users. The reconstruction of the LTE EPC architecture with desired settings can be easily accomplished by using available models. With the help of this simulation experiment can be performed. EPC (Evolved Packet Core) is the framework of Long Term Evolution which provides data and voice on 4G LTE is used. Considering the topologies by peer to peer and use according to the networks.

Examining and configuring the different voice codecs by sending them on voice traffic. Normal and error prone are the two scenarios on which the different four voice codecs is tested. And in the last performance analysis is have been evaluated. Moreover model errors are those error models which suggest the packets lost by error in the network.

In Error based on the specified pattern will be found for empirical model packets. For example Model error listing. This paper considers under the used to evaluate the efficiency of the LTE network at various error rates in terms of packets. Furthermore on off software with an on application has been used to send VoIP traffic via LTE to different codecs. It is an on/off pattern traffic generator, in order to identify the period off and state. During "off" and constant state no traffic is created. Continually delivering flow, which may be described by the "package size", "data rate" offered.

The LTE network is using the preset packet size and data rate for codecs. Application on-off and this setup is relied on the codec type used which is used.

The module of Flow monitor was used for computing the considered QOS parameters. This module tracks network traffic flow and can analyze system capacity. In order to control packet flow, tests must be set up in network nodes, as well as testing the parameters at which time loss is transferred and received in the first/last packet flow to which they belong (characteristic of the probe). On account of VoIP (non-constant stream) PF has a lower proportion than in different schedulers comparative with the other bundle losses [32][33][34][35][36][37].

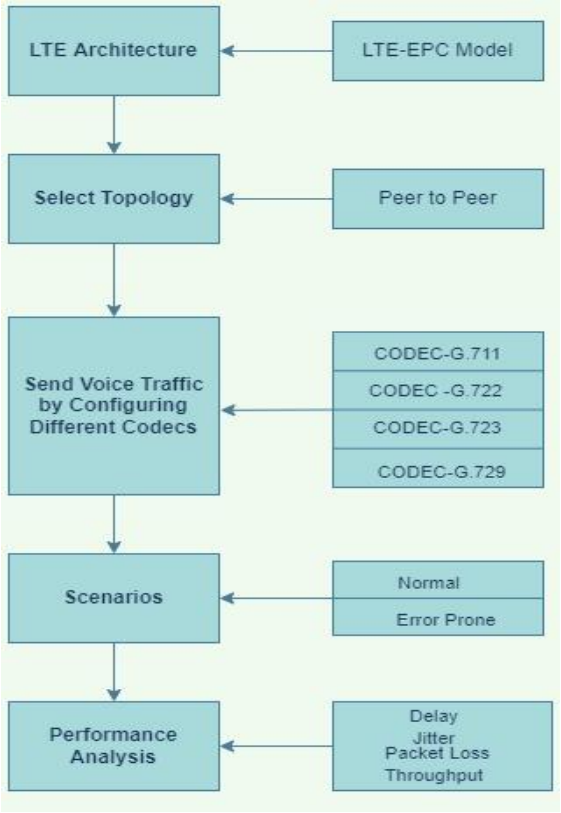

Figure 3 Complete setup of simulation process

EXPERIMENT AND RESULTS:

Experiment is held on MATLAB using simulators for different codecs and QoS parameters are also examine. Moreover it will focuses on how much the data is transfer in given time and the time taken by the data packet for transferring over the network.

Codecs**:**

Codecs is transform compressor, which transmitted hand, uncompressed form, different of which described International Telecommunications Union rules (ITU-T). Sound or the bandwidth of each unit is different. Certain codecs have a better compression ratio, which means less bandwidth and more calls are allowed. Higher bandwidth usage and less call support less calling. All compression-based codecs today are loss, although the use of such codecs reduces voice quality to ensure that voice transmission requires less bandwidth[33]. For reference, focus on two G.711 and G.729 voice codecs with 84kbps, respectively 8kbps bit rate.

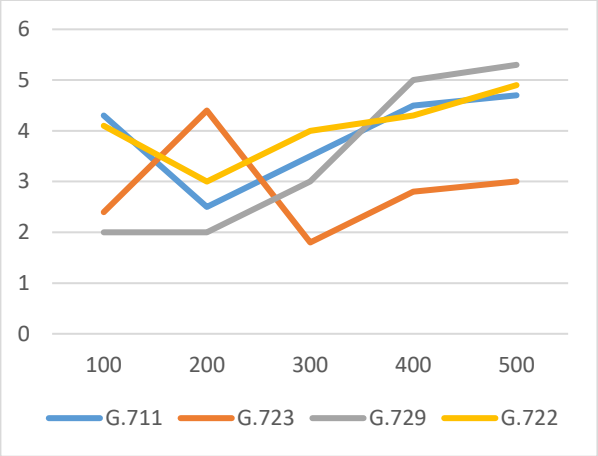

Figure 4 Comparisons of codecs

As shown in the figure 4 and table 1, the delay for somewhat with each deferral, and the blunder inclined organization for the G.723.1 codec shows more fluctuation in mistake inclined climate is clear contrasted with other codecs and in mistake inclined organization. G729 codec shows more accuracy.

Today, the most critical part of voice over Internet Protocols is supremacy. The important evaluation is in terms of lower latency and appropriate throughput for monitoring the efficiency of LTE networks. The voice over IP accuracy of 4G LTE was analyzed and discussed. QoS parameters include results, latency and jitter. Lower calls than are stronger. Some VoIP calls are differentiated from the networks by delivering a fixed rate of output means that produces a steady output stream unaware of the state of the network, requiring additional channel bandwidth, creating congestion that results.

The most popular VoIP according to the suggestions of the ITU-T G-series is shown in table 1. Good assessment and input to them. Proposed model to achieve good results a compare to past year researches as shown in the table 1 below.

Table 1 Results of Codecs Experiment

Type of Codec	Transfer rate	Packet Size	Size of Frame
G711	84 kbps	100 bytes	20 ms
G722.2	45.84 kbps	80 bytes	40 ms
G723.1	9.3 kbps	44 bytes	50 ms
G729	6 kbps	08 bytes	15 ms

MME uses signaling messages to monitor high-level operations such as setup of the bearer, mobile protection, etc. it only transmits for any packet flow in the LTE network. In VoIP, different applications may need different codecs, the bandwidth is restricted and wireless channels are therefore more likely to make mistakes such as interference, the packets may be lost during network transmission.

This paper needs to take into account study of four codecs in the standard a "zero" because it has a stringent need for analyzing. On the basis of this assessment, it is simple to pick the voice codec. The original LTE version has a higher rate of 300 Mb/s. A latent radio network of less than five ms. The capacity of a radio network increases dramatically compared to earlier mobile networks. LTE provides a broad range of network bandwidths and a wide range of different band provisions. LTE network at various error rates in terms of packets. On off software: With an on application has been used to send VoIP traffic via LTE to different codecs. It is an on/off pattern traffic generator, in order to identify the period of and state. During "off" and constant state no traffic is created. Continually delivering flow, which may be described by the "package size", "data rate" offered. The LTE network is using the preset data rate and packet size for codecs.

Table 2 Results of Quality of Service

Number Of Parameters	Results
	U-E-1
Number Of Hops	E-node8-1
UL & DL Bandwidth	30 MHz
Scheduler	PF
Time of Simulation	9.6 sec
Error Model	REM

The Quality of Service is basically is the overall performance of the services of the LTE network. The number of hops is the total number of network devices or the total number of networks from which the data have been passed from the source to the destination. UL & DL stands for Upload Link and Download Link of the bandwidth which is measure in MHz. An algorithm PF stands for Proportional Fair is used for balancing and maintaining two things. First thing to be ensure that all users receive at least basic level of services and second thing for maximizing the total network throughput. The time taken in simulation is measure in seconds. The number of error packets which loss in the network is consider in error model as shown in the table 2.

Feature 1– Normal:

This segment was compared and evaluated at the "zero" errors in the same simulation environment. The flow monitor measures the efficiency metrics. The result shows comparing codecs used for VoIP with scenarios in a normal environment with various techniques. The popularity of VoIP is increased as the science advances. The two technologies that gives us opportunities to transport voice on the internet are the VoIP or the IP telephones and can be used in global communications systems, such as WIFI, LTE and Skype Demands. The Flow Monitor module was used for computing the considered QOS parameters. This module tracks network traffic flow and can analyze system capacity. This paper considers under the used to evaluate the efficiency of the LTE network at various error rates in terms of packets.

Feature 2 – Including errors:

The present paper describes the abilities of the error-prone network. The QOS parameter analysis was performed taking various error rates and measurements into account. The LTE network at various error rates in terms of packets. Moreover On/off software an application has been used to send VoIP traffic via LTE to different codecs. It is an on/off pattern traffic generator, in order to identify the period off and state. During "off" and constant state no traffic is created. Continually delivering flow, which may be described by the "package size. Any codec has been delayed at the destination node. The delay for somewhat with each deferral, and the blunder inclined organization for the G.723.1 codec shows more fluctuation in mistake inclined climate is clear contrasted with other codecs and in mistake inclined organization.

CONCLUSION:

In this paper, examination on various kinds of codecs is held that can be utilized with ns-3 test system in VoIP over LTE. As far as deferral, jitter, parcel misfortune and error-proposed scenarios, the study of the different VoIP codecs across the LTE network has been completed. The VoIP seems to have a great influence from both social and scientific environments. It also analyzes all aspects of voice communications quality over IP networks that have a significant effect. The first is to look at the benefits and the weaknesses. In any case, there are various sorts of codecs that lift voice streams dependent on transfer speed and application necessities. There is, still, no default codec reasonable to everybody. Some codecs have better effectiveness, with different conditions. The collection is based on the comparison of different results of experiments. It would allow customer loyalty with good quality of services and applied in overall network. Providing simulated which may consider being more practical mobile. Efficient network control frameworks can be demanded to boost QOS.

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