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Performance Analysis of Voice over Internet Protocol(VOIP) Application over LTE using various codec schemes

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ABSTRACT
The WiMAX framework successfully underpins wide assortment of broadband remote access advancements, which including rapid web and sight and sound access with high caliber of administration (QoS) prerequisites. Constant administrations, for example, Voice over Internet Protocol (VoIP) are getting to be well known and are real income workers for system administration suppliers. Notwithstanding, there are still numerous difficulties that should be tended to give an enduring and great quality voice association over the best exertion WiMAX and MANET system .To bolster adaptability, productivity and different prerequisites of QoS over a scope of various applications and situations a few provisioning and instruments are given . This examination work explores and examination the execution of Voice over Internet Protocol (VoIP) movement utilizing IPV4 and IPV6 over WiMAX and MANET systems and the effect of different voice codec plans and measurable conveyance for Voice over Internet Protocol (VoIP) over WiMAX has been researched in point of interest. Through different reenactment tests under practical systems administration situations, this study gives an understanding into the Voice over Internet Protocol (VoIP) execution in the WIMAX and MANET systems. The recreations results show that better decision of voice codec's and measurable circulation have noteworthy effect on Voice over Internet Protocol (VoIP) execution in the WiMAX and MANET systems and Performance of chose parameters will be done utilizing the system test system OPNET Modeler.

Keywords: VoIP, QoS, OPNET, MANET, WiMAX,

I .INTRODUCTION

For flexible correspondences organization suppliers (CSPs), Voice over Long Term Evolution (VoLTE) has ascended as the favored response for engaging steady voice action in the rising universe of all-IP frameworks. When this move is done, the circuit-traded framework for voice trades (in which a conferred circuit way is held for each call) will be supplanted by an all-IP framework in which stream based QoS instructs framework resources of worth necessities to ensure call quality. The motivation to thoroughly move to this all-IP approach is strong: until CSPs can reinforce steady IP voice advantages that meet the lifted prerequisites for call quality and trustworthiness set by ordinary circuit-switch frameworks, they will be bothered with the goliath capital and working expenses of keeping up two separate frameworks. As a result of this business driver, VoLTE use is expected to take off: before the end of 2014, there are required to be 59.6 million VoLTE participations set up, and it is evaluated that around 56 percent of LTE-related cell enrollments will use VoLTE organizations before the end of 2019. Voice over LTE is the thing that happens when your bearers grants you to put a phone bring over your LTE affiliation as opposed to the more typical voice frameworks. Verizon Wireless, for occasion, uses 1XRTT for most of your voice calls and LTE for data.

II.VOLTE OVERVIEW

Voice over LTE is the thing that happens when your transporters permits you to put a telephone bring over your LTE association rather than the more normal voice systems

(i)The Benefits o f VoLTE

VoLTE For both supporters and framework managers, VoLTE offers genuine advantages.The study conveyed the going with bits of information:

• VoLTE call quality essentially surpassed that of 3G circuit-traded voice and was quantifiably higher than the HD voice organization offered by Skype4

• With framework stacking (i.e., clusters of battling development), and particularly with establishment applications running on the cell phone and trading data with the framework, the VoLTE results were broadly better than anything Skype

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• VoLTE call setup time was about twice as fast as 3G Circuit Switched Fallback (CSFB) call setup
• VoLTE used essentially less framework resources than Skype voice, which in this way realized longer assessed contraption battery life for the supporter and a more compelling framework for CSPs
• When leaving LTE scope, VoLTE conveys were successfully offered over to 3G circuit-traded voice, ensuring calls continued without impedance

(ii) General architecture

In the VoLTE the most basic part is the VoLTE UE, LTEcomponent including E-UTRAN and EPC, the IMS part including CSCF: Call Session Control Function that havethree component P-CSCF (P for Proxy), S-CSCF (S for Servin)and the application server that join Telephony and SMSapplication server. The figure.1 exhibit the VoLTE building

(iii) VoLTE principal Nodes Description

The down to earth centers of the VoLTE building are described by 3GPP and are depicted underneath. Extra information can seen in 3GPP TS 23.002

a. VoLTE UE (User Equipment)

The User Equipment that is used to take up with the EPC, in the figure 1bove this is a LTE capable UE getting to EPC by method for the LTE-Uu radio interface.

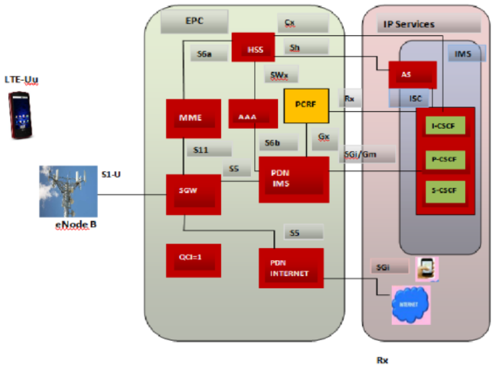

Fig.1: VoLTE Architecture

b. Evolved Universal Terrestrial Access Network (E-UTRAN)

The EUTRAN involves a singular center, the eNodeB that interfaces with the UE.2.3.3 Evolved Packet Core.

c.MME (Mobility Management Entity)

The Mobility Management Entity (MME) is the key controlnode for the LTE access framework.. The MME gives the control plane ability to convenientce amongst LTE and 2G/3G access frameworks and interfaces with the home HSS for meandering UEs.

c. SGW (Serving Gateway)

The SGW courses and advances customer data packages, while in like manner going about as the adaptability catch for the customer plane in the midst of intereNodeB handovers and as the hook for transportability amongst LTE and other 3GPP progressions (finishing the S4 interface and giving off the movement between 2G/3G systems and PGW).

d. IMS

IMS is the control establishment for supporting front line IP Multimedia Services and involves various diverse segments which are recorded underneath.

e. P-CSCF (Proxy Call Session Control Function)

The P-CSCF is the fundamental motivation behind contact for session motioning for the IMS-enabled VoLTE UE. The P-CSCF carries on as a SIP mediator by sending SIP messages between the UE and the IMS Core Network, keeps up the security relationship amongst itself and the VoLTE UE, and joins the Application Function a portion of PCC to enable definitive of the IMS session with the transporter for applying dynamic game plan and tolerating notification of transport level events. The P-CSCF may be realized in an Access Session Border Controller which may in like manner join the IMS-ALG/IMS-AGW.

f. I-CSCF (Interrogating Call Session Control Function)

The I-CSCF is the contact point inside an overseer's framework for all affiliations bound to a customer of that framework. On IMS selection, it flame broils the HSS to make sense of which sensible S-CSCF to course the requesting for enlistment. For adaptable closure calls, it interviews the HSS to make sense of which S-CSCF the customer is enrolled on [2] [4]

g. S-CSCF (Serving Call Session Control Function)

The S-CSCF surrenders session set, session tear-down, session control and guiding limits. It makes records for charging purposes for all sessions under its control, and summons Application Servers in perspective of IFCs got from the HSS. The S-CSCF goes about as SIP recorder for VoLTE UEs that the HSS and I-CSCF dole out to it. It request the HSS for the fitting endorser profiles and handles calls including these end concentrates once they have been enlisted [2].

h. Telephony Application Server (TAS)

The TAS is an IMS Application Server offering support to a base course of action of required MultiMedia Telephony MMTel) organizations as portrayed by 3GPP.

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III.LITERATURE REVIEW

A.Demers et. al.(1989)In this paper, the maker has presented Maestro, an information controller that can manage resources on gigantic plate bunches to give execution detachment among various applications. Maestro screens the execution of each application and effectively allK2ots the display resources so that distinctive execution essentials can be met without static separating. It supports diverse execution estimations (e.g., stillness and throughput) and application needs so that basic applications improve execution if there ought to be an event of benefit question. Maestro makes it possible to use stockpiling resources beneficially by ensuring that high-require applications offering stockpiling to various applications gain the execution levels they require. The trials show that Maestro can constantly change the segment of circle display resources for finish application execution targets.

B. Gumenyuk et. al.(2009) This paper is committed to the issue of effect diverse components on the rate of data transmission in Wi-Fi arranges and growing efficiency of devices in such kind of frameworks. Hence specific work of different rigging was likely investigated. Also proper programming and standards of a remote framework for giving sensible organization was inspected. Besides, purpose of this paper is to give quick and dirty examination and examination of essential execution issues related to remote frameworks in light of the IEEE 802.11b and IEEE 802.11g measures.

D. Hashmi et. al.(2006) IEEE 802.11 standard is basically comprehensively used for giving remote frameworks organization ability to a collection of usages. In spite of the way that the standard has been created basically to give data correspondence limit, it is transforming into a standard mechanical assembly for voice exchanges as well. The clarifications behind this interest are two fold; one is in all cases openness of WLAN frameworks and the other is the insignificant exertion of making voice brings over this framework. Regardless, transmitting voice development k2over WLAN does not utilize the open information transmission (BW) capably and this has nudged a huge amount of activity in the locale of improving the voice call limit. This paper proposes an arrangement for growing utmost by offering need to the passageway point (AP)

Iana Siomina et. al.(2008) In this paper, the maker has discussed the capacity of a best effort framework for different organizations in multi-organization LTE frameworks as to breaking point of a LTE framework with customers at the same time running diverse services.The maker has shown a structure generation study for two-organization circumstances with VoIP and a second organization addressed by constant video, adaptable TV, or web surfing close by results for a besk2t effort orchestrate and had stood out from those for a framework with QoS provisioning. The paper demonstrates that action division and organization prioritization are particularly dire when a delay essential organization, e.g., VoIP, is in mix with a deferral savage concentrated development

Hannes Ekstrom et. al.(2009) In this article, the author has portrayed the QoS thought of the propelled group structure, that had been systematized in 3GPP Release 8. This idea gave access framework heads and organization executives with a course of action of contraptions to enable organization and supporter partition. Such mechanical assemblies are ending up being continuously basic as heads are moving from a singular to a multi-organization offering meanwhile as both the amount of versatile broad band supporters and the development volume per endorser is rapidly extending.

Haythem Assem et. al.(2013)ITU-T proposition G.107 exhibited the E-show, a repeatable way to deal with assess if a framework is set up to pass on a VoIP call or not. Distinctive studies show that the E-model is brain boggling with various components to be used as a piece of checking purposes. Along these lines, reworked variations of the E-model have been proposed to unravel the figurings and focus on the most basic segments required for watching the call quality. In this paper, we propose fundamental solution for an unraveled E-model; we exhibit to figure the change coefficients for 4 customary codecs (G.711, G.723.1, G.726 and G.729A) and thereafter we show that its desires better match PESQ scores by completing it in a checking application.

Haythem Assem et. al.(2013)We portray a testing framework that can give online evaluations of sound and video call quality on framework courses, without requiring either end-customer consideration or prior openness of sound/video game plans or framework takes after. The framework consolidates a mechanical assembly that mimics the sound and video action of IP gets and uses an extended E-Model to evaluate the sound quality

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and VQM to gage video quality. Also, it can mirror framework incapacities to run tests in different framework conditions. Our examination results exhibit that the quality estimations acquired using the framework balance well with the most typically associated industry standard for target voice and video disengaged from the net testing-PESQ and PSNR independently.

Kewin O. Stoeckigt et. al.(2010)Remote voice over web tradition (VoIP) is a crucial rising organization in telecom in light of its potential for supplanting cell telephone correspondence wherever a remote neighborhood (WLAN) is presented. Late thinks, regardless, prescribe that the amount of voice calls that can be maintained in the extensively passed on IEEE 802.11 WLAN is obliged. In this paper, we utilize an affirmed transmission opportunity (TXOP) parameter of a medium access control (MAC) tradition as a fundamental response for upgrade as far as possible. We give a point by point logical model to show that the breaking point can by and large be improved and discuss the consequences of the TXOP parameter to the extent the most great number of calls the IEEE 802.11 framework can reinforce. The informative results are endorsed by multiplications for a broad assortment of parameters. Besides, we inquire about the impact of the pad at the passage point (AP) on the amount of reachable voice calls. We show that there exists a perfect pad size where the most great voice cutoff is expert, however energize extending the support past this value won't realize an extended voice limit. In perspective of this finding, a close structure expression for the most compelling number of voice calls is made as a component of the TXOP regard. Finally, we propose a direct yet exact conjecture for voice-limit estimation and give a couple of bits of information got from the theory.

IV.SIMULATION ENVIRONMENT

Table 1: Simulation Parameters for WiMAX

Cell Radius	30km
No. of Base Stations	7
No. of Subscriber Stations per BS	10
Speed of the mobile nodes	50, 100, 150 m/s
Simulation time	600 sec
Base Station Model	wimax_bs_ethernet4_s lip4_router
Subscriber Station Model	wimax_ss_wkstn
IP Backbone Model	ip32_cloud
Voice Server Model	ppp_server
Link Model (ASN - Backbone)	PPP_SONET_OC12
Physical Layer Model	OFDMA 20Mhz
MAC Protocol	IEEE 802.16e
Multipath Channel Model	ITU Vehicular A
Traffic Type of Service	Interactive Voice and Data
Scheduling Type	ertPS, nrtPS
Application	IP telephony
Voice Codec (with and without silence suppression)	G 711, G.729, G.723, G.726,G.728
Inter repetation time	Constant 200

A Voice Codec utilized at the client side to change over the simple voice waves into advanced heartbeats and the other way around. There are different codecs sorts in view of the picked information rate, inspecting rate, and executed pressure calculation portrayed in table.2

Table2:Various Codecs

Codec	Coding rate
G.711	64
G.722	48
G.723	5.3 AND 6.4
G.726	16, 24
G.727	16 TO 40
G.728	16
G.729	8

(i)Scenario 1 – Static Nodes

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Figure 2 portrays the reenactment setup used for WiMAX network.Employing the OPNET Wireless Deployment Wizard a 7 cells WiMAX system, with a few client stations in the base station reach is conveyed. The base station is connected to the center system by a server spine through anIPbackbone.

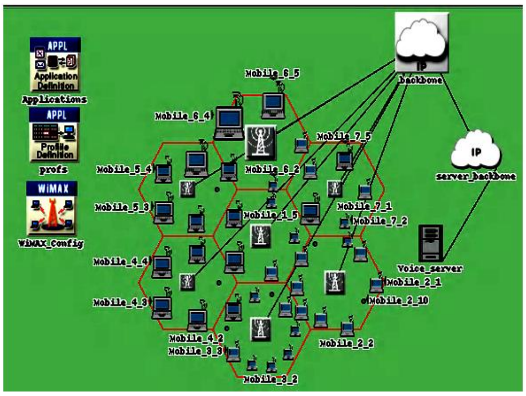

Figure 2. Network Model for WiMAX in OPNET

A. Average Jitter

Figure.2 delineates the adjustment in jitter for the WiMAX system without using quiet concealment for a few codecs. Watched voice quality is ideal if the jitter is zero. As portrayed in the figure, mean voice jitter is almost 0 for the voice codecs G 723.1 with both 5.3Kbps and 6.3Kbps and G726 with 32Kbps importance great quk2ality of voice while all different codecs shows some deviation.

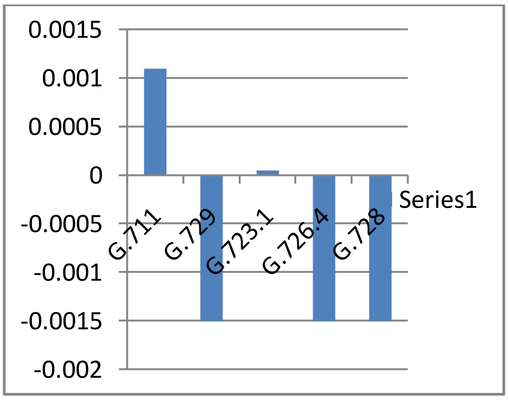

Figure 3. Average voice jitter without Silence Suppression

To expand the quantity of clients upheld number, quiet concealment instrument is noteworthy. This is depicted in Fig 4. In this manner, G. 726 can't be utilized in situations where quiet concealment system is used paying little respect to its execution in situations where hush concealment has not been utilized.

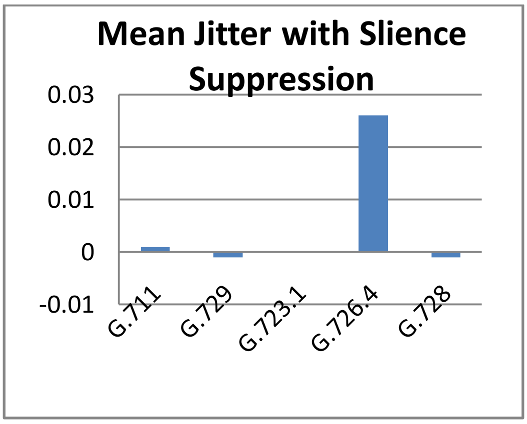

Figure 4.Average voice jitter using Silence Suppression

B. Average Packet End to End Delay

Bundle systems takes a shot at parcel exchanging standards; hence voice in a WiMAX would be transported to the destination hub as an arrangement of parcels where each one may receive different courses, along these lines reach at the destination with different deferrals. Variables affecting bundle end to end delay include Packetization Delay, Look Ahead Delay, Network Delay, Serialization Delay, and so on. Figure 5 and 6 shows, the Packet End-to-End delay for the voice codec G 723.1 is the most elevated paying little heed to quiet concealment. This is expected; G 723.1 uses coding rate of 6.3 Kbps or 5.3 Kbps which results in the developing of bundles of bigger tally and littler size. Presently with expanding number of parcels in the system, the blockage in the system increments. Clog specifically impacts the system parcel delay and in this way brings about expanded bundle end to end delay.

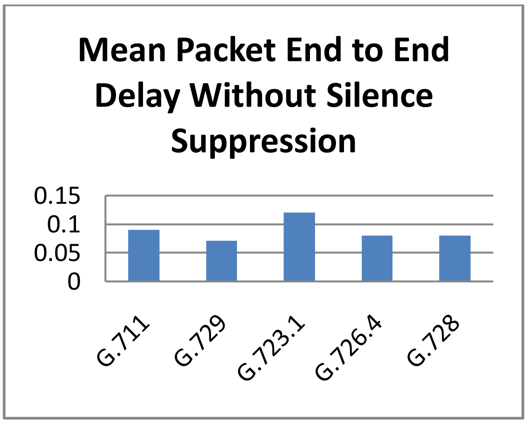

Figure 5.Average packet end to end delay without Silence Suppression

Additionally the Look Ahead deferral of G 723.1 voice codec is 7.5 msec [25] though the

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same for G 729A is 5 msec and other accepted voice codec's is 0 msec. Then again the serialization deferral of G 726 32Kbps is fairly high.

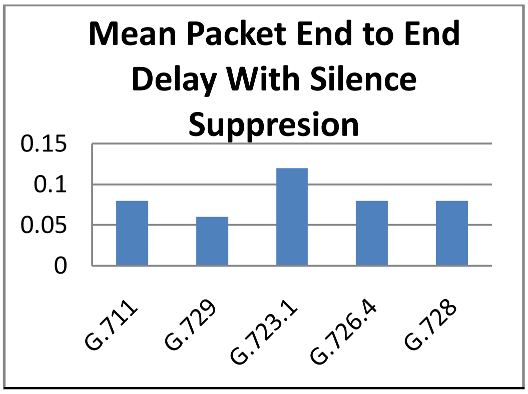

Figure 6.Average packet end to end delay using Silence Suppression

C. Average MOS

The Mean Optimal Score (MOS) as showed in Fig 7 and 8 is not reliant of Silence Suppression. MOS relies on upon number of bundles lost. G 723.1 is a low piece rate codec which makes parcels of size 6.3 Kbps or 5.3 Kbps. This outcomes in system clog and along these lines parcel misfortune. In this way, the MOS esteem for the voice codec G 723.1 is somewhat low. Voice having MOS of 3 can be thought to be of impressive quality. Therefore, all other codec's extensive as for their MOS esteem.

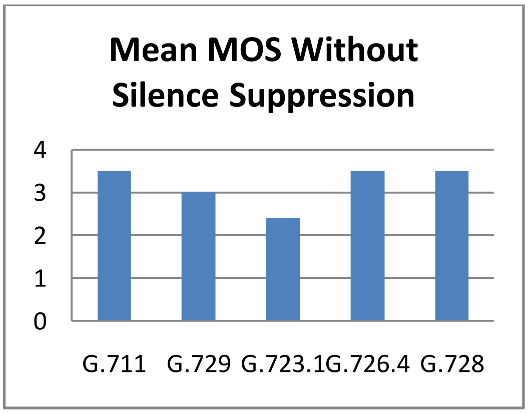

Figure 7. Average MOS without Silence Suppression

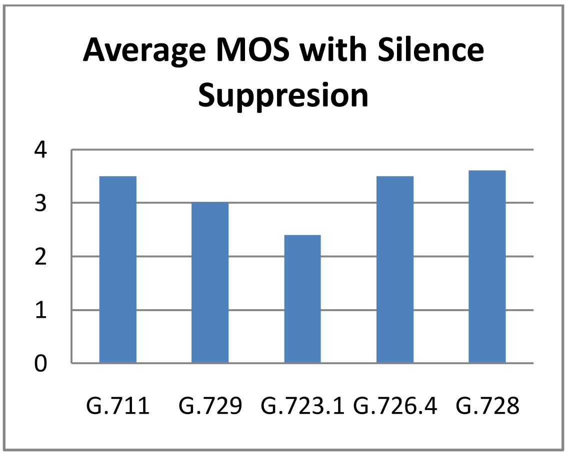

Figure 8.Average MOS with Silence Suppression

(ii) Scenario 2 – Mobile Nodes

Fig 9 depicts the reproduction setup used for WiMAX system. Utilizing the Wireless Deployment Wizard of OPNET a 7 cells WiMAX system, with different endorser stations in the base station reach is conveyed. The base station is connected to the center system by a server spine through an IP spine.

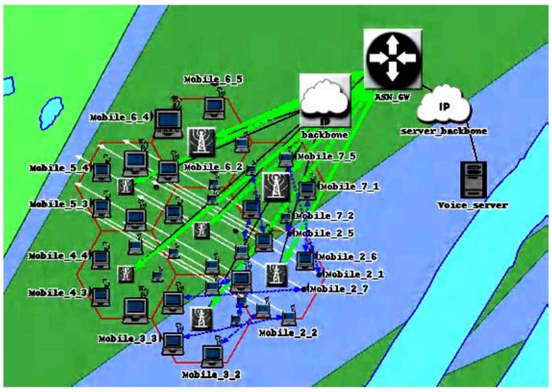

Figure 9. Network Model for WiMAX

A. Average Jitter

G 711 codec has bundle rate of 64 Kbps which is very expansive in correlation of G 723.1 which is almost 5.3 Kbps. Thusly, G711 has less number of parcels in correlation of G 723.1 for settled measure of voice. For bigger parcels the overhead due to header is little [28].

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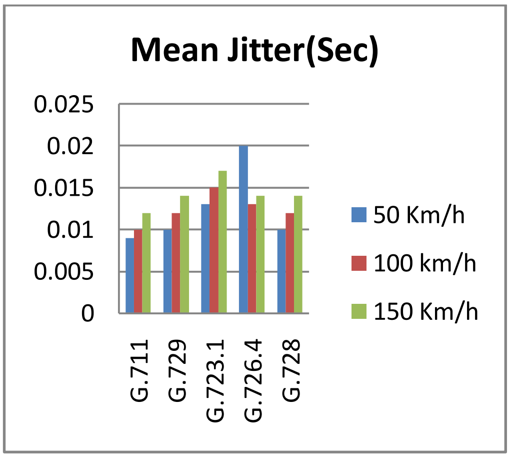

Figure 10.Mean Jitter without Silence Suppression

Thus, for situation 1, the deferral for G 711 is lesser in examination of G 723.1. moreover, in situation 1, the hubs are static in this manner the bundles follow just about the same way and scope all together pretty much yet remote systems, have a basic property of disregarding any bundles containing one or all the more wrong piece.

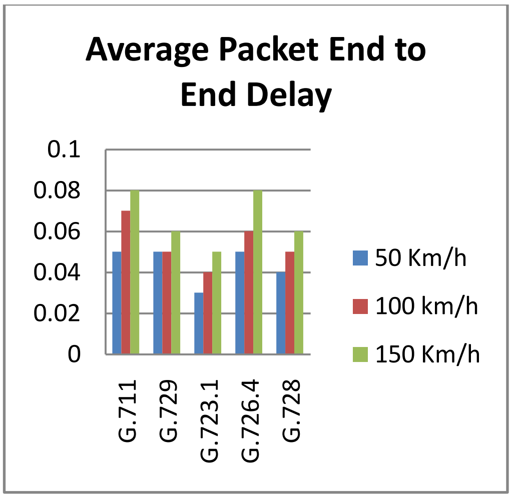

Figure 10.Mean Packet end to end delay without Silence Suppression

So the likelihood of disposing of G723.1 parcels are less and the jitter is additionally minimum for this codec. Still in the portable situation, situation 2, the likelihood of bundle drop is increasingly and any parcel drop in the physical layer is deciphered as postponement by the upper layers in view of the disguising property of the connection or MAC layer conventions [28]. Since, the parcel size of G 711 is bigger; it has higher probability of misery **from** bundle drop when contrasted with G 723.1 therefore having higher deferral in correlation of G 723.1. This is shown in Figure 4.9.Since, the quantity of parcels for G 723.1 is a great deal more when contrasted with G 711, this issue is more hazardous when contrasted with G 711. Subsequently it experiences most elevated jitter.

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