Performance Evaluation of 3G internet access

Ayodeji Akeem, Ajani
Department of Electrical and Computer Engineering,
Kwara State University, Nigeria
ajaniaa@gmail.com

Rowani Odum
Department of Data Telecommunications and Networks,
University of Salford, United Kingdom
r.odum@edu.salford.c.uk

Muheeb Olanrewaju Ahmed
Department of Electrical and Computer Engineering,
Kwara State University, Nigeria
muheebahmed@yahoo.com

ABSTRACT

Internet technology is advancing with increasing internet applications demands. However, most developing countries have poor internet experience. Thus, the Quality of Service, QoS of an internet technology, Third Generation, 3G in a developing country, Nigeria was studied. Voice over internet protocol, VoIP and Video Streaming were the internet applications while CODECs and node density were the choice of the performance parameters. Thus, the performance of the 3G network (simulated in Qualnet 6.2) was evaluated based on QoS performance metrics (Average End-to-End Delay (s), Average Jitter (s), Throughput (bps) and Mean Opinion Score). The results showed 3G network can deliver reasonable QoS for VoIP depending on CODEC choice and that node density has adverse effect on QoS. The problem identified for Nigeria was total reliance on mobile broadband and high node density. Upgrading of 3G network, migration to 4G network and revival of fixed networks were recommended to improve QoS.

CCS Concepts

• Networks → Network performance evaluation → Network performance analysis

Keywords

3G, Simulation, Qualnet 6.2, QoS, Internet access, VoIP, CODECs, and Node Density.

1. INTRODUCTION

The recent global internet data rate increment can be attributed to the improvement in broadband technology over the years. Broadband technology can either be fixed or mobile. Fixed Broadband is the internet access provided through cabled network while Mobile Broadband refers to broadband internet access obtained through mobile cellular networks. [1]

There seems to still be a gap between the QoS experienced in developed country and the developing ones despite the global improvement. For instance, a case of the United Kingdom (a developed country) against Nigeria (a developing nation). The average broadband speeds reported by OOKla [2] at the time of the research were 29.9 Mbps and 5.9 Mbps for the UK and Nigeria respectively. Another important statistic showed that UK [3] primarily depend on fixed broadband while Nigeria depends mainly in mobile network for her internet access [4]. This implies that the internet experience between the two countries is widely apart. This inspired the need to evaluate the performance of mobile broadband in providing internet access. Despite the deployment of 4G (with theoretical target peak data rates of 300Mbps and 75 Mbps on the downlink and uplink respectively) [1] most of the developing country such as Nigeria are still widely using 3G network. The full migration from 3G broadband to 4G broadband might take longer than expected in Nigeria as the country broadband vision for 2018 is to achieve 3G network coverage in 80% of the country [5]. This research used 3G network as its reference network.

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The aim of the research was to evaluate the performance of 3G broadband internet access for households in terms of service experienced by users and make recommendation on issues learnt during the study for better performance and improved end users' experience using Nigeria as the case study.

The following objectives were set in order to achieve the aim; Present an overview of 3G network for broadband internet access; Review past papers related to 3G broadband internet access to residential users; Design the 3G broadband network.; Implement the design in simulation using Qualnet 6.2 simulator [6]; Evaluate the quality of service performance of network using different internet applications; Critically analyze the result of the simulation; Suggest possible solutions to the problem discovered in the case of Nigeria; and Present a clear conclusion based on lessons learnt during the research and recommendations on areas not covered.

2. BACKGROUND

This section presents the overview of 3G broadband internet access, 3G architecture and its Quality of Service, QoS.

2.1 3G broadband Internet Access

Universal Mobile Telecom System, UMTS is a Third Generation, 3G telecommunication system which offers advanced data services (broadband internet) in addition to the primary telephony service. 3G internet access is classified as broadband [7] as it uses Wide Code Division Multiple Access (WCDMA) which involves wider bandwidth. The World Administrative Radio Congress, WARC assigned a total of 230MHz frequency band around 2GHz spectrum range to be used in 3G network [8].

3G network has improved data rate compared to earlier mobile network and can support new IP services like voice over internet protocol, VoIP and video chat. In addition, UMTS has a Quality of Service scheme employed in the network to support real time application and manage the traffic effectively. There is also improved resource management control in 3G network which assigned to the user resource based on the service requirement. UMTS is also more secure compared to previous generation as it uses two-way authentication technique and data sent over its air interface are also encrypted. UMTS feature and ability were realized from a systematic architecture developed by the Third Generation Partnership Project, 3GPP.

2.2 UMTS Network Architecture

According to Third Generation Partnership Project, 3GPP specifications, [9] UMTS network consist of the Radio Access Network, RAN and the Core Network, CN. The RAN is made up of the Users' Equipment (UE), Node B and the Radio Network Controller (RLC). The CN has both the packet switched and circuit switched infrastructure. The circuit switched is basically based on GSM network while the packet switched evolved from 2.5G infrastructure. The packet switched network has better internet traffic implementation compared to circuit switched network, because of the transport layer capability of handling burst data traffic in the PS network. The PS core network consists of the Serving GPRS Support Node, SSGN, Gateway GPRS Support Node, GGSN and the Home Location Register, HLR.

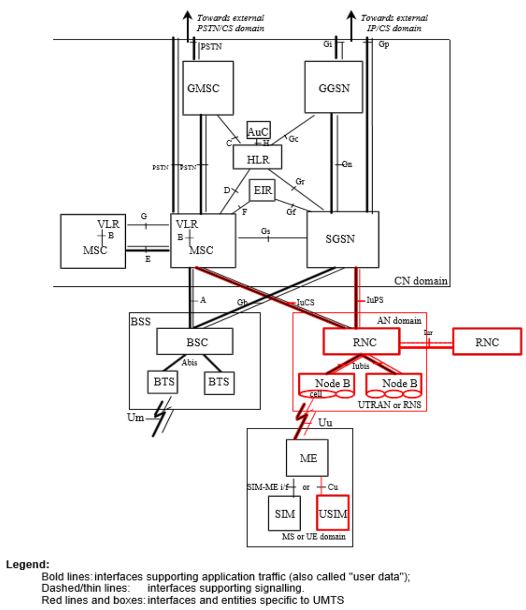

Figure 2.1: UMTS and GSM Network Architecture [10]

The air interface between the UE and the Node B is called Uu interface while the Iub interface connects Node B to RNC. Iur is used for RNC to RNC connection while IuPS and IuCS are used for connecting the UTRAN to the PS and CS core networks respectively.

The architecture was expected to make 3G deliver good CS and PS service quality. This was based on the 3G Quality of Service, QoS requirements.

2.3 UMTS QoS Classes

UMTS traffic has been classified based on quality of service requirement by 3GPP into four (4) classes which are conversational class, streaming class, interactive class and background class [11]. Delay sensitivity is the main QoS metrics used in the classification, with conversational class for traffic with high delay sensitivity and the background class for traffic that has least sensitivity to delay.

Conversational and Streaming classes are mainly intended to be used to carry real-time traffic flows. The main difference between the two classes is the direction of the traffic, Conversational traffic are bi-directional while streaming class are unidirectional. Examples of Conversational class include VoIP and Video Chat while audio and video streaming are common examples of Streaming class. Interactive class and Background are mainly representing traditional Internet applications like World Wide Web (WWW), Email, Telnet, and FTP. The main difference between Interactive and Background class is that Interactive class is mostly bi-directional, while Background class is unidirectional.

3. LITERATURE REVIEW

[12], compared the performance of UMTS release 99 and HSDPA in terms of their capacity and throughput using internet traffic mix. They observed HSDPA had better capacity and also had better throughput. They also noticed that node density had a negative relationship to the throughput.

[13], did performance analysis of Wireless Local Area Network, WLAN standards (IEEE 802.11 a-g) in delivering Voice over IP traffic. Their performance analysis was based on the CODEC scheme and the transport protocol. The CODEC compared in their experiment were G.711, G.723, G.726 and G.729 while Session Initiative Protocol and H.233 were the transport protocol used for their analysis. The performance metrics used for their study were end-to-end delay (s), jitter (s) and Mean Opinion Score, MOS. Their findings revealed the CODEC G.729 and G.726 both had better performance in terms of end-to-end delay and jitter but G.711 had better MOS performance. They also argued H.233 to be a better protocol than SIP when it comes to delivering Voice over Internet protocol. They advised comparative analysis of different coding scheme on other wireless networks such as UMTS, LTE and WiMAX. They argued that CODEC plays an important role in determining the QoS of any wireless network using VoIP application.

[14], did a comparative analysis of performance of WiMAX and UMTS network in delivering Voice over IP traffic, VoIP service. The study simulation in Opnet. The CODEC used for the research was G.711. The performance metrics used for comparison were the packet end-to-end delay (s), Jitter (s), Mean Opinion Score, MOS and Packet Delay Variation (s). They concluded based on their experimental result and analysis that the performance of WiMAX in delivering VoIP service was better that UMTS based on the performance metrics. They also observed node density had negative effect on the VoIP QoS and this was more evident in the UMTS network.

[15], did comparative analysis of QoS performance of UMTS, Wi-Fi and hybrid (UMTS-Wi-Fi) network in delivering Voice over IP service. The study was simulated using Opnet. The scenarios consisted of users making VoIP calls using the three networks. The CODEC used was G.711. The performance metrics used for the analysis were the packet end-to-end delay (s), packet delay variation (s), jitter (s) and Mean Opinion Score, MOS. They concluded based on their experimental analysis that the Wi-Fi network gave the best performance in terms of end-to-end delay and packet delay variation while the hybrid network delivered best in terms of jitter and MOS. They argued that UMTS had poor performance relatively in based on all the metrics used. UMTS had very high delay, jitter and the lowest MOS.

The papers reviewed characterized UMTS with high delay which is poor QoS but [13] argued that the choice of CODEC plays pivotal role in QoS of VoIP. They further proved this using a WLAN network. Two different CODECS were used on UMTS network as presented Section 5.3. Also, the work of [14] and [15] also revealed the node density has negative effect on 3G network performance in delivering internet access using different internet applications like VoIP and CBR. Most of these work did not test to the capacity of the network and did not make technical inference of the situation. Detailed analysis of Node density effect on 3G network will be discussed in Section 5.5 and 5.6.

4. RESERCH METHOD

4.1 Design Requirement

The set up for the research required 3G user(s), 3G network and internet service(s).

4.2 Design Components Description

Sequel to the requirement, setting up a live 3G network or using an existing one for the purpose of the research would have been too expensive and could have negative consequence on the network. Thus, the design was simulated using QualNet 6.2 by scalable technologies [6]. QualNet 6.2 has the full components for 3G networks and its implementation. Also, it has provision for different internet applications.

4.3 Design Components' Configuration

4.3.1 3G components configuration

The UMTS components consist of the User's equipment, UE, Node B, RLC, SGSN, GGSN and HLR. The UEs served different functions in the research depending on the application. UEs were used to make and receive phone calls (in phone call application), make VoIP call (in VoIP application), stream video (in Streaming Application) and access web page (in HTTP application). The parameters configuration discussed in this section was an extract from the entire configuration based on the role each parameter played at different layers. The physical and MAC layer configuration were very essential for the UE, Node B and the connecting air interface between them, Uu. The uplink and downlink channel were configured as 1950 MHz and 2150 MHz respectively. The uplink channel was used for communication from the UE to the Node B while the downlink represents the signal transmission from Node B to UE. The frequencies chosen were based on 3GPP spectrum allocation to Europe and Africa as specified. The antenna for the UE and Node B had similar configuration except for the transmission power and height. The antenna model used for the research were omnidirectional using two-way path loss model. Omnidirectional antenna has the same antenna gain irrespective of direction of the transmitted and received signals. This was chosen to allow even distribution of UEs around Node B for the node density scenario. The simulation area used was 1500 square meters made up of 100 m by 100m square boxes. The coverage area of the Node B was set to maximum distance of 1000m. This was because the research was set to mimic a typical urban settlement. According to NCC, most of the internet access in Nigeria are majorly in the urban settlement. The H323 model was used for the multimedia signaling protocol because it was standardized by ITU-T and has been especially focused on smooth interworking with the PSTN.

The GGSN also requires dual configuration for its interface. The UMTS cellular configuration was used for GGSN interface to other UMTS core network components (SGSN and HLR) while the IP configuration was used for GGSN interface to IP Network. This was necessary for routing packets to and from 3G network. The entire wired link, (Iub, Iur, IuPS, IuCS and IuG) were configured to have abstract MAC layer with size of 10Mbps. The other core network components, HLR and SGSN were also configured using cellular model to allow full functionality of the 3G network.

The IP devices also serve different functions like UEs depending on the internet application. IP devices were used receive VoIP call (in VoIP application), video source (in Streaming Application) and web server (in HTTP application). The IP devices where configured to route packet to and from 3G network through the GGSN.

4.3.2 Internet Applications

The Internet applications used were VoIP, Streaming (represented by Constant Bit Rate, CBR on simulator), HTTP and Background update representing each of the four QoS classes.

VOIP simulates IP telephony in H323 network. The initiator and receiver generate real time traffic with an exponential distribution function. That simulates a real life telephone conversation. Examples of VoIP application include skype and viber applications.

The Constant Bit Rate (CBR) traffic generator generates traffic at a constant rate by transmitting packets (also called “items”) of a fixed size at a fixed rate. It can be used to simulate applications for which the end-systems require predictable response time and a static amount of bandwidth is continuously available for the life-time of the connection. These applications include services such as video-conferencing and telephony (voice services). [6].

The QualNet HTTP model was based on the following standards: RFC1945 Hypertext Transfer Protocol -- HTTP/1.0., RFC2068 Hypertext Transfer Protocol -- HTTP/1.1 and RFC2616 Hypertext Transfer Protocol -- HTTP/1.1. HTTP simulates single-TCP connection web servers and clients.

4.3.3 Phone Call Application

This mimic the conventional CS call on 3G network.

4.4 Performance Parameters

The performance parameters used for the research were CODECS for the VoIP application and the node density for the streaming application.

4.4.1 CODEC

CODEC encodes audio signals into digitized form at the source. This digitized signal is sent to the receiver by using packet based internet. The receiving terminal receives packets and decodes them to transform into audio signal again using CODEC. Based on [13] argument, two different CODECs (G711 and G729) were investigated for VoIP service on 3G network to examine the effect of the choice on the QoS performance of the 3G network.

4.4.2 Node Density

The research is also interested in investigating the effect of node density on QoS performance of 3G network. This was particularly important as Nigeria has many users using the mobile network compared to the UK as analyzed in Section 5.5.

4.5 Performance Metrics

These were the metrics value used in measuring the user's end experience to judge the performance of the network in delivering the internet service to the user. The metrics used are, Average End-to-End Delay (s), Average Jitter (s), Throughput (bps) and ean Opinion Score

4.6 Design Description

The network design was based on the components identification and configuration discussed.

The design has three different groups of scenarios.

The first group was the scenario used to verify and validate the simulation process. This was the foundation scenario which had all the applications used in the design. This scenario consisted of six users' equipment, UEs in the UMTS network and three IP devices connected to the gateway. Two of the UEs were making UMTS call conversation, while the other four UEs were making VoIP call to an IP device (VoIP receiver), browsing HTTP website (from another IP device configured as web server), streaming video (from the and forth UE for background traffic both from last IP device configured as the CBR client. This implementation in QualNet 6.2 is as shown in Figure 4.1.

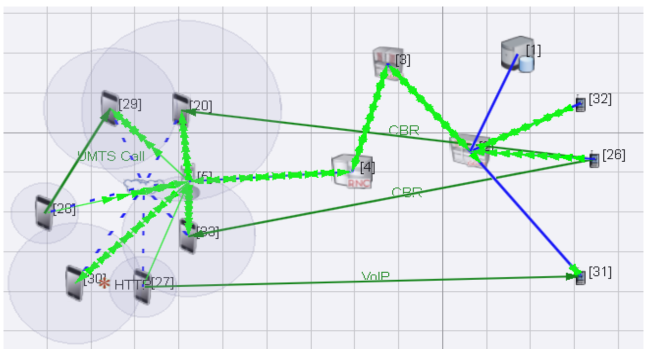

Figure 4.1: Complete design implementation in Qualnet 6.2[8]

The design implemented in Figure 4.1 was inspired by the 3GPP description of internet access over 3G network which is as shown in Figure 4.3. The Node B (node 5 in the picture) was in charge of communication with the UEs while RNC (node 4) assigned the resources needed by UEs at the Node B. The core network components (HLR (node 1), GGSN (node 2) and SGSN (node 3)) were responsible for authentication of the UEs as well as routing of internet packet within the 3G network.

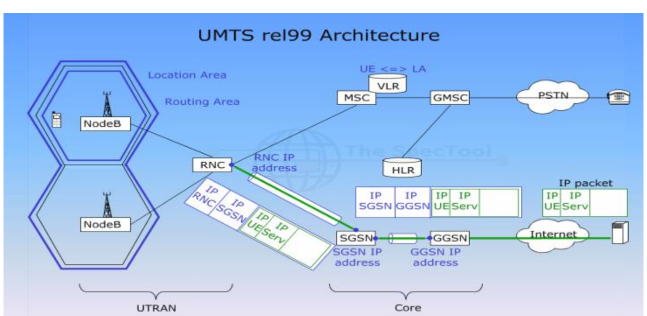

Figure 4.2: 3GPP network design [16]

The second set of scenarios were set to investigate the QoS of 3G network using VoIP application as an example of the conversational class which required the highest QoS. This was important to understand how well 3G performs in delivering internet access. Unlike [14] and [15] who concluded that 3G has very high delay in delivering VoIP application, the research used another CODEC, G.729 to see if there could be a better performance by 3G compared to what they obtained using G.711.

The last set of scenarios investigate the influence of node density on 3G QoS. This has been done using streaming application which also requires high QoS like VoIP but in one direction. CBR application was used for this implementation using different item sizes of 60B, 80B and 100B to generate different download rate of approximately 48,000bps, 64,000bps and 80,000bps. This represented a typical download rate for MPEG 4 [17].

Different download rate with constant Iub size of 10 Mbps implies different number of UEs for each item size. The number of UEs was increased by 10UEs in subsequent scenarios until one or more UEs were unable to stream at the estimated download rate. The size choices were within the range specified by [6].

5. SIMULATION IMPLEMENTATION AND RESULT DISCUSSION

5.1 Verification and Validation Scenario

This was set up was to mimic full implementation of both Packet-switched (PS) and Circuit-switched (CS) applications as described by 3GPP as discussed earlier in Section 4.6.

UMTS call was chosen to represent CS application while VoIP call, Streaming, web browsing and background file download were chosen for PS application using each to represent the four UMTS QoS category as explained in Section 2.3. The simulation period was chosen to be 600 seconds. This simulation period has been maintained throughout the subsequent scenarios. Some of the results behaviours that verifies and validates the simulation are discussed here.

5.1.1 Verification and Validation Scenario Implementation

Table 5.1: Configuration Paramters

APPLICATION	NOTABLE PARAMETERS CONFIGURATION
UMTS Call	Avg. Talk time (10s), Start time (10s), Call Duration (580s)
VoIP	Avg. Talk time (10s), Start time (10s), End time (end of simulation), CODEC (G711), Precedence Value (7)
Streaming	Item to send (random), Item size (60B), interval (0.01s), Precedence Value (4)
HTTP	Start Time (10s), Threshold time (1s)
Background	Item to send (random), Item size (500B), interval (0.01s), Precedence Value (1)

In the streaming application, item size was chosen to be 60 bytes with an interval of 0.01s to give an estimated received throughput of 48 000 bps as obtained in Calculation 5.1.

Item size = 60 bytes = (60 * 8) bits = 4800 bits

Estimated received throughput = item size / interval = 4800 bits / 0.01 seconds = 48 000 bps

Calculation 5.1 Estimated received throughput for streaming application.

The streaming rate is similar to the rate of MPEG 4 which is a video type in 3G UMTS device [5].

Using Calculation 5.1 for the backgroud application gave estimated received throughput of 400 000 bps.

5.2 Result discussion of the Verifiaction and Validation Scenario

The result obtained after running the configured design in QualNet 6.2 verified that there was phone conversation between the two UEs as there was exchange of CS packets between them. The UMTS call application had no value for PS packets since it was not a PS application.

Table 5.2: All applications scenario traffic property at the UE

	VoIP	Streaming	HTTP	Background app.
Total Unicast Data Sent, Received (B)	2.2e+06, 1.9e+06	NA, 3.5e+06	128202, 3.7e+06	NA, 2.6e+07
Total Unicast Messages = Fragments Sent, Received	13923, 11856	NA, 58996	366, 383	NA, 52010

There was VoIP communication as well. It was exchange on PS packets as againgt the CS call and the conversation was biredirectional. The performance metrics showed a high delay and a moderate jitter. This was similar to the values obtained by [14] and [15] which made them conclude that 3G can deliver VoIP service but with high delay. Going back to the requirement of VoIP which is to have a similar QoS behaviour to UMTS call, then the result is a deviation from this. However, [13] argued that choice of CODEC plays important role in QoS of a wireless network and proved this investigating WLAN. Thus, the research investigated this argument in Section 5.5 using 3G network instead to understand the VoIP performance using different CODECs. Also, the number of messages was equal to the number of fragments meaning there was no fragmentation. This is because the message size, fragment size and the packet size were approximately 160B (as obtained in Calculation 4.4.).

Table 5.3: All applications scenario QoS performance metrics at the UE

Performance Metrics	UMTS CALL	VoIP	Streaming	HTTP	Background app.
Unicast Received Throughput (bps)	NA	26175.3	48160.2	50155.4	354359
Average Unicast End-to-End Delay (s)	0.0424	1.7218	0.05233	3.973	14.408
Average Unicast Jitter (s)	NA	0.01028	0.01497	0.130	0.1354
Average MOS	NA	1.28189	NA	NA	NA

Total data received = 1.89696e+06 bytes

Total packet received = 11992

Total message received = Total fragment receive = 11856

Size of message (B) = Size of fragment (B) = Size of packet (B) Total data received (B) / Total message received = 1.89696e+06B / 11856 = 160B

Calculation 5.3 Size of message, fragment and packet for VoIP application.

The video streaming application also showed a one transfer of PS packets as backed by theory. The estimated throughout obtained from estimated throughput and the obtained throughput are both similar as well. The performance metrics showed a very low delay and a low jitter. This showed a good QoS. Also, the number of messages was equal to the number of fragments meaning there was no fragmentation. This was because the message size, fragment size and the packet size were approximately 60B (which can be obtained in using corresponding values in Calculation 5.3.). The message being smaller than the VoIP traffic is also an indication to while VoIP has more delay than the CBR traffic suggesting that a better compression could give a better QoS. The calculated size of message 60 byte is equal to the item size 60 byte configured for the streaming application.

HTTP and background application result details also followed the internet application pattern. The only notable differe is the fragmentation that occurred in HTTP application as the message size greater than 544 bytes. The framents size is 489.4 bytes as derived in Calculation 5.4

Total data received = 3.67527e+06 bytes
Total message received = 383,
Total fragment receive = 7503,
Size of message (B) = Total data received (B) / Total message received = 3.67527e + 06 / 383 = 9596 B
Size of fragment (B) = Total data received (B) / Total fragment received = 3.67527e + 06B / 7503 = 489.8B

Calculation 5.4: Size of message and fragment for HTTP application.

Thus, from the analysis above, the design configuration did not only worked (verified) but also work in line with existing theory and similar experiment (validated).

5.3 3G QoS Investigation Using Voip Application Only

The approach used was based on UMTS Quality of Service Category. As discussed is Section 2.3, UMTS QoS classification was based on the application requirement. The conversational class required the best performance because it involved real time application and also bidirectional which means the same quality is expected in both directions. The application chosen for the research was VoIP because of its increased usage and also because it has been fore-tipped to compete with conventional UMTS call. This was particularly relevant in Nigeria's context because most people find VoIP calls cheaper compared to normal CS call. This was principally significant when making international calls as the VoIP charging is based on internet usage and not location based unlike CS calls which usually have a higher tariff for international calls compared to local calls.

The main QoS target of VoIP was to have a similar or even better performance when compared to the UMTS call. This research set to achieve this using different CODECs. The CODECs used were G.711 and G.729. The choice of G. 711 was because it is the most generally used CODEC and one of the earliest CODEC developed for VoIP while G. 729 was one of the latest developed CODEC for VoIP application.

The ITU describe G.711 CODEC [18] as the CODEC which uses lossless compression scheme in coding speech signal at 64 Kbps while G.729 CODEC uses Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP) in coding speech signal at 8Kbps.

The VoIP application only scenario is similar to VoIP implemetation descibed in Section 5.1.2 except that the G729 was in addition to G711. The talking time was later changed from 10s to 20s and 30s to have another set of randomly generated values. The summary of the VoIP application only scenario is summarised in Table 5.4.

Table 5.4: Configured parameters for VoIP application scenario only

VoIP Initiator, VoIP Receiver	UE, IP Device
AVERAGE TALKING TIME (s)	10, 20 and 30
START, END TIME (s)	10, 0
Call status	Accept
CODECs	G 711 and G729
Packetisation, interval	Interval, 20ms
Priority, value	Precedence, 7

5.4 VoIP Application Only Scenario Result Analysis

The speech coding for both G.711 and G.729 were calculated to be approximately 64Kbps and 8Kbps respectively for every scenario. This was compared to the 64Kbps and 8Kbps specified by ITU. This was done for the purpose of validation.

Total Data sent = 2.10224e + 06B = 1.681792+7b

Talking time = 263.962s

Speech coding rate = 1.681792 + 7b / 263.962s

= 63713.4 bps

Calculation 5.5: Speech coding rate for G.711

Total Data sent = 322220B = 2577760b

Talking time = 322.942s

Speech coding rate = 2577760b / 322.942s = 7982.11 bps

Calculation 5.6: Speech coding rate for G.729

The result obtained from the VoIP application only implementation as extracted in Table 5.5 showed a better performance using G.729 compared to G.711. The range of average delay of G.729 (0.024 to 0.0287s) was low (similar to CS voice call with average delay of 0.0424s in Table 5.2) compared to the G.711 (1.1967 to 2.5190s). The same imporvement was noted for Average Jitter and MOS.

Table 5.5: Results obtained for the VoIP Scenario

CODEC	G. 711			G.729		
Talking Time (s)	10	20	30	10	20	30
Average Unicast End-to-End Delay (s)	1.196 7	1.659	2.519 0	0.024 0	0.026 8	0.02 87
Average Unicast Jitter (s)	0.010 2	0.013 8	0.019 9	0.000 1	0.000 1	0.00 01
Average MOS	1.448 1	1.233 7	1.132 7	3.270 0	3.270 6	3.26 38
Unicast Received Throughput (bps)	27528 .6	30230 .1	35156 .5	3513. 1	4314. 4	485 8.9

The reason for its better performance was because it has better compression for the messages. The size of a message in G.729 is 20B compared to 160 B in G. 711.

Table 5.6: Result properties for the VoIP scenario

CODEC	**G.711**	**G.729**
Average Talking Time (s)	10	10
Total Unicast Data Sent, Received (B)	2.1e+06, 2.0e+06	322220, 255480
Total Unicast Messages Sent, Received	13139, 12501	16111, 12774
Number of PS data packets sent, received	13252, 12617	16224, 12894
Average UL, DL radio resource allocated at Node B (bps)	133645, 133479	133601, 133514

Total data received = 255480B

Total packet received ≃ Total message received = 12774

Size of message (B) ≃ Size of packet (B) = Total data received (B) / Total message/ packet received

= 255480B / 12774 = 20 B

Calculation 5.7 Size of message and packet for VoIP application using G.729

Summarily, G.729 offered better QoS performance in terms of average end to end delay, jitter and MOS as it has better compression technique than G.711. The only drawback is the throughput, though not a problem from the user's view as their experience was better but on the part of the network provider, this was significant because the same average resources were applied at Node B to process both traffic. The importance of better utilisation of resources is to accommodate more users with the available resources. The result of the simulation conformed to [20] argument. As a result, it was believed that a 3G network can deliver a good QoS provided the right CODEC choice has been made. The investigation of node density effect on 3G performance will be discussed in next section having concluded on the VoIP single user investigation.

5.5 Investigation of Effect of Node Density On 3G QoS using Streaming Application

Studying the effect of node density of 3G internet access perfomance was particularly important comparing the approximate node density of the research reference countries, Nigeria and UK. The node density is the number of users per cellular base station. Nigeria has approximately 4403 users to a base station while the UK has approximately 1143 users to base station as derived in Calculation 5.8.

Total number of cellular subscriber in Nigeria = 131 910 228 [4].

Total number of cellular base station in Nigeria = 28 289 [4]

Total number of cellular mobile users in UK = 40 000 000 [3]

Total number of cellular base station in UK = 35 000 [3]

Estimated node density of Nigeria ≃ 4662

Estimated node density of UK ≃ 1142

Calculation 5.8: Cellular node density for the UK and Nigeria

Another indication that the QoS problem in Nigeria could be node density is that Nigeria despite having high number of users to base station, Nigeria has more internet users than UK and most of the internet users depend on the mobile network unlike UK who has both fixed and mobile network.

The node density implementation for the research was done using streaming application only. The choice of streaming application was because it has a predictive performance metrics, throughput (bps). The streaming application has an expected guaranteed rate for the download. This will help identify the point network fails to deliver to expection. Unlike the VoIP whose throughput is always randomly generated. Also streaming class requires good QoS like VoIP. Another reason for choice of streaming application over VoIP was that it is unidirectional. This was particularly important as the UMTS implementation on QualNet 6.2 was based on Release 99 which has upload limitation. The problem of upload limitation has been taken care of in the 3G later release, HSUPA.

The implementation of streaming application only scenariosis similar to the implementation in Section 5.1.3. In addition to 60B item used in Section 5.1.3, items of sizes 80B and 100B were used to generate an estimated data rates of 64,000bps and 80,000 bps. The summary of the streaming application only scenarios implementation is presented in Table 5.7. The scenarios for each item size had users in increment of 10.

Table 5.7: Different scenarios implemented for node density scenario

Parameters	**Value**		
Item Size (B)	60	80	100
Number of UEs	10 to 110	10 to 100	10 to 90

The scenario started with 10 UEs and it was increased for each item size until the scenario where one or more UEs were receiving very small or negligible throughput.

5.6 Streaming Application Only Scenario Result Analysis

The result analysis for all the scenarios were based on the throughput because the throughput can be estimated for CBR applications. The result analysis was built on single UEs situation for each item size. 48063.1bps, 64084bps and 80128.9bps were obtained throughput for single UE using item sizes of 60B, 80B and 100B respectively.

The amount of resources allocated by RNC at Node B for single UE were obtained from simulation as 86796bps, 101147bps and 116873bps for 60B, 80B and 100B item size respectively. The difference between the allocated resources and the received unicast throughput caters for the control packets and forward error correction, FEC. The number of data packet for all item sizes were obtained as 59000 packets. This was because the packet transfer was assigned to send an item in 0.1s for 590s (600s (simulation time) - 10s (start time)). This will give a total of 59000 packets (590 / 0.01) as obtained. Also, the number of control packets applied by the RNC was obtained as 30003 packets. This implied that each UE will have resources assigned to cater for the data packets, control packets and the FEC. This analysis of the allocated resources was very important in order to account for the Iub size which was set as 10Mbps.

The node density investigation for the research started with 10 UEs with increment of 10 for the subsequent scenarios. This was done for the item sizes chosen.

It was observed from the result obtained that all the UEs had the estimated throughput of approximately 48000bps for the 60 B scenario up to 80 UEs scenario.

The total received throughput by UE for this scenario was calcluated as 3.84Mbps (80 * 48 000bps) while the total allocated resources by RNC was of 7.4 Mbps of 10Mpbs assigned.

However, scenario with 90 UEs for 60B item size showed some UEs were unable to get up to the estimated throughput of 48000bps. The effect of the node density was clearly being noticed at this stage of the simulation. The situation of 90 UEs were similar to the 100 UEs as the number of UEs with reduced throughput increases. The scenario series for 60B item ended with 110 UEs as some of the UEs had negligible throughput.

Thus, the analysis of the 60B item set of scenario was done using 10 UEs to 100UEs. The allocated resources for the 100 UEs scenario was 9.2 Mbps of 10Mbps assigned.

Similarly, it was observed from the result obtained that all the UEs had the estimated throughput of approximately 64000bps for the 80 B scenario up to 70 UEs. The total received throughput by UE for this scenario was calcluated as 4.4Mbps (70 * 64000bps) while the total allocated resources by RNC was of 7.2 Mbps of 10Mpbs assigned. However, scenario with 80 UEs for 80B item size showed some UEs were unable to get up to the estimated throughput of 64000bps. The effect of the node density was clearly being noticed at this stage of the simulation. The situation of 80 UEs were similar to the 90 UEs as the number of UEs with reduced throughput increases. The scenario series for 80B item ended with 100 UEs as some of the UEs had negligible throughput.

Thus, the analysis of the 80B item set of scenario was done using 10 UEs to 90UEs. The allocated resources for the 90 UEs scenario was 9.4 Mbps of 10Mbps assigned.

Also, It was observed from the result obtained that all the UEs had the estimated throughput of approximately 80000bps for the 100 B scenario up to 60 UEs. The total received throughput by UE for this scenario was calcluated as 4.8Mbps (60 * 80000bps) while the total allocated resources by RNC was 7.002 Mbps of 10Mpbs assigned. However, scenario with 70 UEs for 100B item size showed some UEs were unable to get up to the estimated throughput of 80000b. The effect of the node density was clearly being noticed at this stage of the simulation. The situation of 70 UEs were similar to the 80 UEs as the number of UEs with reduced throughput increases. The scenario series for 100B item ended with 90 UEs as some of the UEs had negligible throughput.

Thus, the analysis of the 100B item set of scenario was done using 10 UEs to 80UEs. The allocated resources for the 80 UEs scenario was 9.6 Mbps of 10Mbps assigned.

Summarily, the number of UEs was increased to maximum of 100, 90 and 80 for the 60B, 80B and 100B item sizes respectively.

Having understood the pattern of the scenario implemented for node density, the analysis of the QoS performance metrics is as follows;

The values obtained were used to plot graph of average unicast received throughput per UE as shown in Figure 5.3.

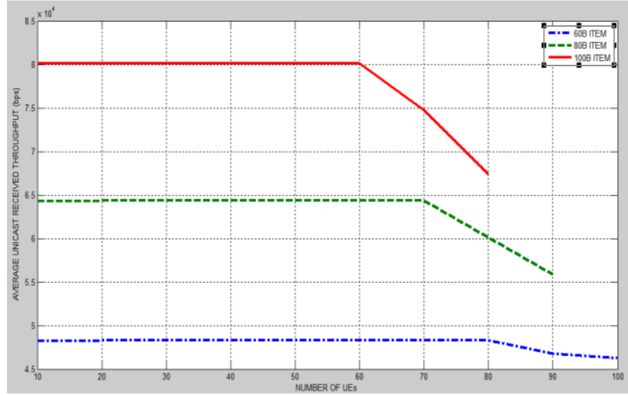

Figure 5.3: Average Unicast Received Throughput per UE (bits/second) for different Item sizes

The graph of the throughput (ploted in MATLAB) gave a constant throughput for some number of UEs. This is the normal behaviour expected because the traffics have been generated using CBR which normally generates at constant rate. However, at some point where the node density (Nd), was approaching the network limit in terms of available resources, the throughput dropped gradually with increase in number of UE.

The values for average end-to-end delay, Dee per UE were obtained and plotted as shown in Figure 5.4.

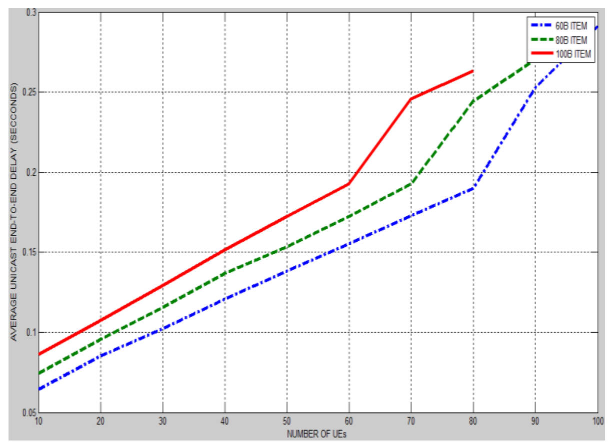

Figure 5.4: Average Unicast End-to-End Delay per UE (seconds) for different item sizes

It was observed that the Dee was directly proportional to the item size, Si. This was because the total size of items processed increases with item size. For instance, the scenario for 10 UEs sees the 3G network processing 600B size (60B * 10UEs) of item for 60B item size scenario but this get higher for 80B item size scenario with 800B size of item (80B * 10UEs) while 100B size item scenario has the highest of the three as 1000B (100B * 10UEs). It was also observed from the set of results obtained that the average end -to-end delay, Dee was increasing at constant rate with increase in node density, Nd (Dee = K*Nd) until the normal point where all scenarios experienced sharp increase in the delay. This is because as the number of UEs increases, the number and size of packets to the processed increase as well.

It was also observed that the average jitter, J was increasing at constant rate with increase in node density, Nd (J = K*Nd) until the normal point where all scenarios experienced sharp increase in the jitter. This is as plotted in Figure 5.5.

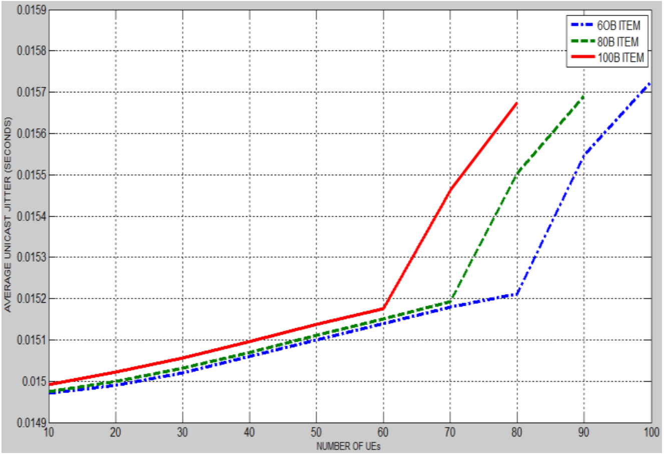

Figure 5.5: Average Unicast Jitter per UE (seconds) for different item sizes

Summarily, the node density have a negative effect on the 3G QoS in terms of the average unicast received throughput, average end-to-end delay and average jitter.

The throughputs generated were within the range specified by 3GPP as 34Kbps to 384 Kbps for one way video streaming. The work of [14] and [15] did show that node density has negative effect on QoS using different applications.

The scenarios established in this section showed that 3G network can deliver good QoS for it users provided all the conditions are right. The instance used in this research was VoIP application which requires a very small delay and jitter. The research achieved good QoS for 3G network using G.729 CODEC instead of G.711. Also the effect of node density was investigated on QoS as this was one of the major problems facing Nigeria's internet.

6. POSSIBLE SOLUTIONS TO QoS AND NODE DENSITY ISSUE IN CASE OF NIGERIA

This Section discuss the possible ways to improve the situation as highlighted in the research aim and objectives.

6.1 Improving the 3G Network

The research believed that if all or most existing 3G networks in Nigeria can be upgraded, a higher capacity and better quality will be delivered to the users. Depending on 3G networks only might not be sufficient for a country like Nigeria. Other possible solutions which can be complementary to the one discussed above will be discussed in Section 6.2 and 6.3.

6.2 Migration To 4G

Advantages of migrating to 4G include but not limited to: presenting users more QoS as well as better capacity and coverage. Hence it is an important consideration for Nigeria to have a proper plan for 4G mobile broadband in their vision for better internet access for her residents.

6.3 Revival of Fixed Internet Broadband

As shown in Calculation 5.8, UK has better node density on cellular network and also have viable fixed network. It will also help both the QoS and node density if Nigeria can work on improving their fixed network.

The three recommendations will improve internet QoS and will also share the users thereby reducing the effect of node density.

7. CONCLUSIONS AND RECOMMENDATION

The research evaluated the performance on 3G internet access in order to suggest ways to improve the internet access in developing countries like Nigeria as highlighted in the research aims and objectives. 3G network was simulated using QualNet 6.2 as described in Section 4. The QoS was investigated using VoIP application and the result showed a better performance for this using G.729 CODEC compared to G.711 as discussed in Sections 5.3 and 5.4. Also, the analysis in Section Sections 5.5 and 5.6 also showed that node density has negative effect on 3G QoS in terms of average end-to-end delay, average jitter and throughput. Three recommendations (which are improving the 3G network, migration to 4G network and having a fixed broadband as a complementary service) were proposed to improving internet accessing Nigeria in Section 6.

There were some technical constraints (related to systems and simulator) during the research. This limited the coverage of the research in terms of its absoluteness. Also, it was difficult getting real life data for implementation by telecommunication industries as they attached commercial values to this.

Comprehensive assessment of the state of infrastructure for different broadband technologies in Nigeria and the economic impact of implementation of each is strongly recommended in order to make a more informed decision on moving forward towards achieving a good QoS for her residents.

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