INTERNATIONAL JOURNAL OF SOFTWARE ENGINEERING & COMPUTER SYSTEMS (IJSECS)
ISSN: 2289-8522 e-ISSN: 2180-0650
VOL. 8, ISSUE 1, 53 – 68
DOI: https://doi.org/10.15282/ijsecs.8.1.2022.6.0096

ORIGINAL ARTICLE

A QoS Estimation Algorithm from Caller Ringtone Analysis in GSM Network

O. A. Ayo-Bello[1] , A. M. Aibinu[2], O. C. Ubadike[3], A. J. Onumanyi[1] and M. O. Momoh[3]

[1]Department of Telecommunication Engineering, Federal University of Technology, Minna, Niger State, Nigeria
[2]Department of Mechatronics Engineering, Federal University of Technology, Minna, Niger State, Nigeria
[3]Faculty of Air Engineering, Air Force Institute of Technology, Kaduna, Kaduna State, Nigeria

ABSTRACT – Call Setup Time (CST) is one of the key performance indicators (KPIs) that Mobile Network Providers (MNPs) are mostly appertain. It has been established that long CST usually severely affects the user experience. Owing to the limitations associated with gleaning the CST data from MNPs, this paper provides the development of QoS estimation algorithm from various CST parameters. The algorithm involves the determination of CST; Inter-Burst time; Intra-Burst time and Call duration in time domain. The caller Frequency content was also determined by the application of fast Fourier Transform before computing the Mean Square Error (MSE). The eventual QoS rating is done after the computation of the MSE from various individual parameters. Four hourly data consisting of 10 sets each were collected three times in a week for four weeks for each MNP's for creating Caller Ringtone dataset and testing the developed algorithm. Performance analysis of the system in accurately determining: CST; Intraburst time; Interburst time and Call durations were carried out. Results obtained shows that the proposed technique accurately computes these parameters and maximum error obtained was to the value of 10%. Furthermore, the QoS obtained shows an error margin of less than 5 % was observed when the developed technique was compared to the ground truth. Thus, the proposed algorithm was able to compute the QoS using Caller Ringtone only, thus independent of MNP.

ARTICLE HISTORY
Received: 12 Dec 2021
Revised: 27 Feb 2022
Accepted: 8 April 2022

KEYWORDS
Quality of Service
Call set up time
Mobile Network
Ringtone
MSE

INTRODUCTION

Global System for Mobile Communication (GSM) network performance and QoS evaluation is an important requirement for MNPs operation. It directly impacts revenue generation and customer satisfaction [1,2]. QoS measurement is a means for controlling the performance, dependability and usability of a telecommunications service. QoS is however affected by several factors in mobile network. There are several criterions that can be used to measure QoS from the user point of view. These metrics includes the coverage, accessibility, and the audio quality [3]. In coverage the signal strength is measured using test equipment which can be used to estimate the size of the cell. Accessibility deals with establishing that the network has ability of handling successful calls from mobile-to-fixed networks and from mobile-to-mobile networks. The audio quality measurement involves monitoring the clarity of a successful call for a period of time for the communication channel; in this work, we considered caller ringtone. All these pointers are used by the telecommunications industry to rate the QoS of a network [4]. Consequently, immense effort is made by most MNPs to monitor their respective networks and maintain accurate status of its quality [4,5]. This status, along with new traffic demand data is used by these operators to improve their network and adjust its operation.

Due to the high demand for GSM services, there has been a decline in the QoS experienced by most end users[5,6]. This has resulted in both external and internal interest in monitoring the level of QoS maintained by most MNPs as documented by[7]. By external, reference is made to independent bodies like the Nigerian Communications Commissions (NCC), and other research interests, apart from the Mobile Network Providers themselves (Internal). However, to measure QoS, it has been the practice to depend solely on data hosted and provided by the MNPs. These data are most often unavailable to the public, except through officially authorized means or some cooperation. Also, fidelity and integrity of such data remain unguaranteed. Therefore, external measurement of QoS status remains a challenge [8].

Consequently, recent trends have sought new methods to measure QoS independent of the MNP data. These have been used in different reports, and have formed a recent area of research. There are several Key Performance Indicators (KPIs) used for measuring QoS such as Call setup time (CST), Call Completion Rate (CCR), Call Drop Rate (CDR), Call Handover Success Rate (CHSR) and Standalone Dedicated Control Channel (SDCCH), but focus here is on the use of CST mean square error for QoS estimation.

*CORRESPONDING AUTHOR |M. O. Momoh | momuyadeen@gmail.com

RELATED WORK

As demands on mobile communication increases, the ability of MNPs to plan and manage capacity, and to deliver superior service to customers, is being severely challenged. Owing to increasing number of users and several smart devices and applications, MNPs have been experiencing some challenges in the areas of improved coverage, cost, channel capacity and QoS [9-11]. A problem in hardware, transmission, coverage, or interference may result in an increase in the call setup time. A faulty Transceiver or combiner or a wrongly connected RF cable makes seizing of the SDCCH or TCH complex, and thus resulted in call setup time been increased. Call Setup Time (CST) refers to the time for an end-to-end call to be set up by MS through radio network equipment.

It has been established that long CST usually severely affects the user experience. Hence, CST is one of the key performance indicator that MNP’s are most concerned about as it provides a measure to regulate and guarantee the QoS requirements[7,12]. Author in [13], described and evaluates real call set up success rate in GSM and also states the possibility of its implementation using the current technologies in GSM and difference between the real and implemented CSSR. The “real” as the Kollár explains means that CSSR is calculated as ratio of the assigned traffic channel (TCHs) to the channel requests. It was concluded that more complex formulation which utilized the Immediate Assignment Success Rate (IASR), TCH Assignment Success Rate and SDCCH Success rate must be used for measuring CSSR. Kollár further stated that this formulation was the best approach despite a higher effort on the processor part of the equipment where the CSSR is to be calculated. It was noted that the formulation did not cover the case when the Direct TCH Assignment feature is enabled.

Author [1], gave insight into the causes of call setup failures in a GSM service test area was studied and the use of RF optimization process to increase call success rate was presented. The adopted methodology is drive testing, post processing and data analysis. The authors conclusion were that most of the network problems are caused by increasing subscribers and the changing environment and also that RF optimization should be carried out frequently in order to improve the network performance with the existing resources [14-15]. CST can be estimated by drive test through the calculation from L3 messages. For GSM network, CST is the period from Requesting a Channel until Alerting, it is usually 7-8 s for Mobile-to-Mobile Calls. The CST measurement can be achieved through traffic measurement and Drive Test (DT). However, based on the need for self-determining measurement, we examine the DT approach, which guarantees independence in the next section[5,16].

Drive test

The procedure used in Drive Test involves using a car containing mobile radio network air interface measurement equipment that can detect wide range of the physical and virtual parameters of mobile cellular service in a given test region. Coverage, capacity and QoS of a mobile radio network can be evaluated with Drive Testing method. Drive test equipment typically collects information and services running on the network such as voice or data, radio frequency scanner and GPS information to provide location logging[2,17]. Drive tests is aimed at tracing the signalling on the Um interface and on the Aibs interface in different scenarios hence, the signalling process can be analysed comprehensively and the problem can be located easily[9,18-19]. There are diverse types of tools to carry out a DT among which are: JDSU E6474A v15.2, TEMS Investigation, Nemo Outdoor. However, in this work, an algorithm was developed using MATLAB and LabView to estimate QoS from the analysis of caller ringtone in GSM network. The result gotten will then be used in estimating QoS as explain in the next section.

METHODOLOGY

The methodology used in this research is divided into 4 blocks each block set to achieve each objective. A test area was selected; Data were collected every 4 hours i.e., 2am, 6am, 10am, 2pm, 6pm and 10pm three times in a week i.e., Mondays’ Wednesdays’ and Saturdays’ for 4weeks and for each hour 10 data (caller ring tones) was collected. The caller tone dataset was then Pre-processed and analysed using the developed algorithm and bench mark against results obtained from manual determination of the caller tone parameters. A standard caller tune file for Globacom was downloaded from the internet as ground truth caller tune. The file was converted with the format converter to a wav file and was fed into the developed algorithm to extract the call parameters for computation of necessary QoS. These parameters were then use as a standard against our dataset to compute the MSE as well as QoS estimation of MNPs base on the difference between the ground truth (downloaded tone) and our recorded tone for selected tone 1-10. In estimating the Qos of a caller tune, the Euclidean Distance Accuracy Measure has been used. The Euclidean distance measures the closeness between the determined caller tune parameters and the ground truth.

Direct listening

The ring tones were listened to by few people one after the other to choose the best of each 10 set. The chosen tone was fed into the design to extract the relevant call parameters such as (silent mode time (SMT), InterBurst Time, IntraBurst time, Call Duration, and Number of Burst). Also, some other parameter like mean squared error, Amplitude of the burst, frequency, Signal-to- noise ratio (SNR) etc. were used for further analysis.

Globacom standard

These MNP has their SMT to be 7 seconds for calls between their customers, it has a standard pattern and sound which completely differs from other network hence the ring tone that best suit this standard was choose from the set. The chosen tone was then used as an Observe Value (y) in QoS estimation.

Test area

Since the aim of the research was to assess the QoS provided by mobile operators as perceived by the user, areas where these services are mostly used were selected, i.e., Suburban Districts of Abuja; these districts are not within the Federal Capital City (FCC) but because of their proximity to the FCC, they have attracted some development and many people who work in the FCC live in these suburban districts. For the purpose of this research Lugbe district is chosen. Lugbe; under Abuja Municipal Area Council (AMAC) is one of the fast-growing suburban settlements in Abuja. It is largely residential and densely populated. Lugbe is divided into five districts namely Lugbe south, Lugbe north, Lugbe central, Lugbe west and Lugbe east. Lugbe has been brought into lime light because of its proximity to the city centre and to the Abuja airport. This made significant development to be attracted to the area.

Heavy traffic is experienced in the early hours of the morning between 6.00am to 10.00am and 4.00pm to 8.30pm on the lanes from Lugbe to city centre and city centre to Lugbe respectively. Journey to or from Lugbe to city centre could take well over one hour during these periods

Proposed QoS estimator algorithm to estimate QoS from caller ringtone in GSM network

QoS estimator from caller ring tone in GSM network is being proposed as part of ongoing research using a citizen sensing approach. This section only focuses on the discussion of the proposed algorithm for QoS estimation which consists of four distinct stages as shown in Figure 1.

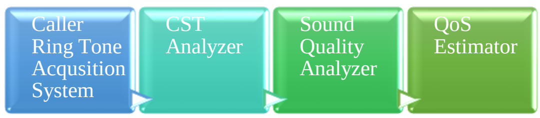

Figure 1: Block diagram of research methodology

This technique is based on the analysis of GSM ringing pattern, call time and the sound quality of the ringing tone. The data collected were converted using the format converter from 3gp format to wav format and fed into CST analyser algorithm. CST analyser then extracts all the relevant call parameters which were used in the analysis of the sound quality and QoS estimation. The call parameters extracted includes SMT also known as CST, Intra-Burst Time, Inter-Burst Time, Number of Burst and Ringing Duration as shown in Figure 2.

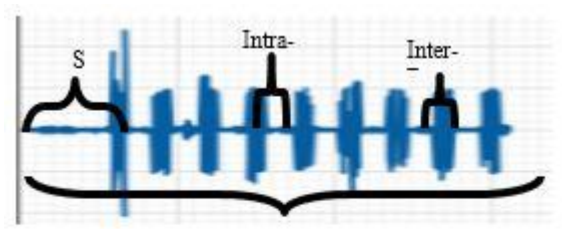

Figure 2: Typical GSM ring pattern

The technique also allows the caller ring tone parameters to be analysed and their mean square error (MSE) was used with respect to SMT to estimate the QoS of the MNPs under observation. Figure 3 shows the flow chart for the caller ringing tone acquisition.

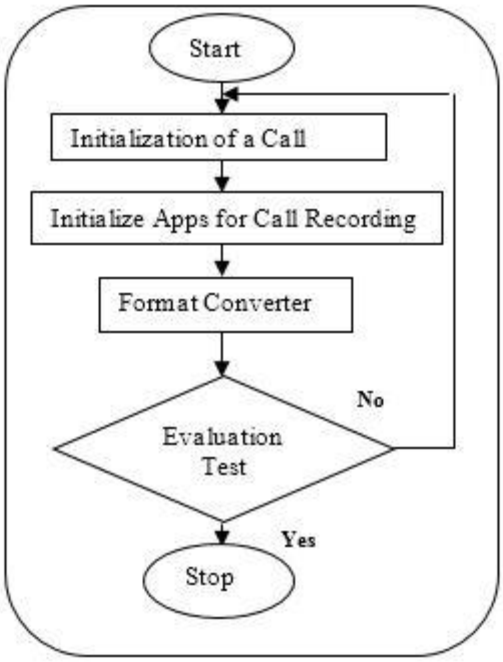

Figure 3: Caller Ring Tone acquisition flow chart

From the CST Analyzer shown in Figure 4, Silent Mode Time (SMT) can be computed mathematically by finding the difference in the start time of the counter from its end time.

$$SMT = t_{end} - t_{start} \tag{1}$$

Where t_{end} is the silence period end time and t_{start} denotes the start time of the silence period. Also, Intra Burst Time (IntraBT) can be estimated by finding the difference in the time the first peak is detected from the time the end of a burst is detected.

$$IntraBT = t_2 - t_d \tag{2}$$

Where t_d and t_2 is the start and end of burst respectively.

Value for Inter Burst Time (InterBT) can be gotten from subtracting the end time of the detection first burst from the time the next burst started.

$$InterBT = t_2 - t_{start} \tag{3}$$

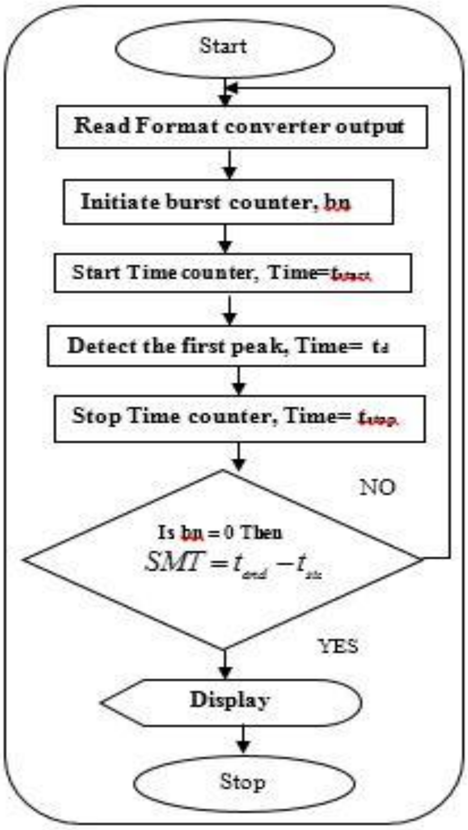

Figure 4: CST Analyzer flow chart

The process involved in analysing sound quality includes Signal in Noise and Distortion (SINAD) measurement, Amplitude measurement and Frequency measurement. This is carried out by following four distinct stages as shown in Figure 5.

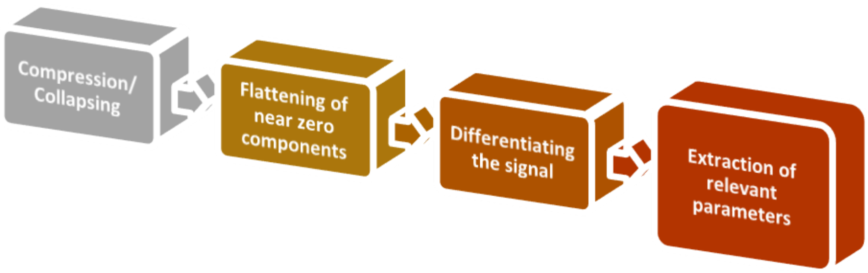

Figure 5: Sound Quality Analyzer block

It was observed that the input signals have sudden and infrequent zero values occurring due to high sampling frequency of 8000Hz per samples and this affect the computational procedure. However, averaging method was used to annul the presence of these zero values in the signal i.e., we were able to effectively compress 8000 samples per second to 8 samples per second without distorting the signal. It was also observed that values that are less than 0.01 are noise; thresholding method was then used to reduce them to zero. The resulting signal is then differentiated to facilitate the effective tracking of gradients. A positive gradient indicates the start of a burst and the return of the gradient to zero denotes the end of burst and the start of an inter burst time. The algorithm used for sound quality analysis for calls originating from different mobile network providers to other mobile network providers in a fairly quiet environment is shown in Figure 6 while Figure 7 is the labview design of the sound quality analyser.

Step 1: - Initialize counter; y, z

Step 2: - Set a threshold =x

Step 3: - Measure start time of the input signal =Tstart

Step 4: - Detect the Amplitude 'A' of the input signal

Step 5: - Check if A > x else Repeat Step 5

Step 6: - Measure T=Tstop

Step 7: - SMT= Tstart - Tstop

Step 8: -Check if A <= x else Repeat Step 8

Step 9: - Measure T=TQ

Step 10: - IntraBurst Time= TQ -Tstart

Step 11: - Counter y=1

Step 12: - InterBurst Time= Tstop - TQ

Step 13: - Counter Z=1 GOTO Step 5

Step 14: - Stop

Figure 6: Sound quality analyzer algorithm.

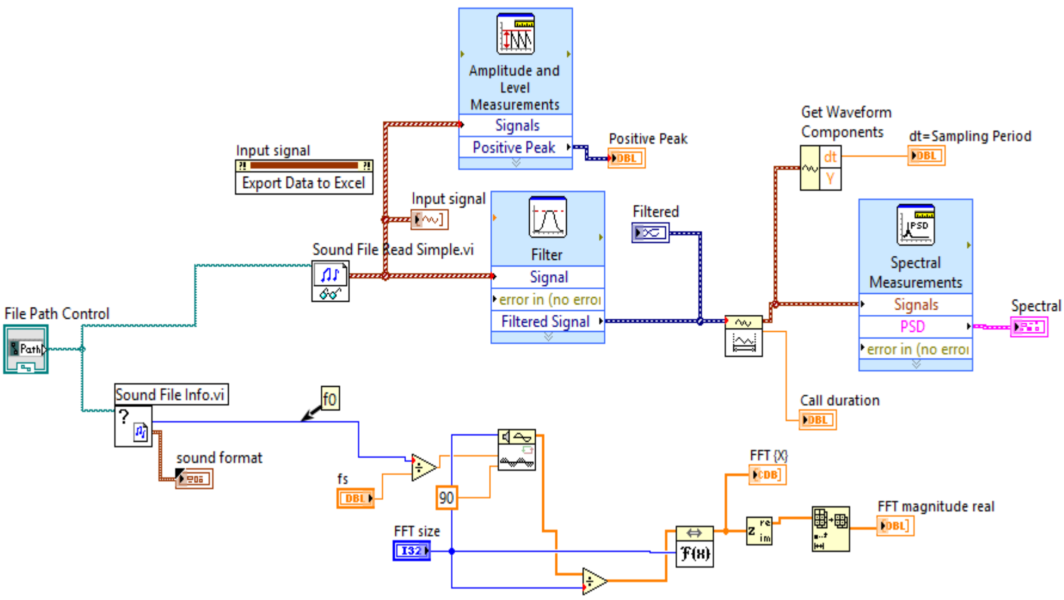

Figure 7: Labview design of the sound quality analyzer.

Preliminary results and discussions

In this section, the caller tone dataset was analysed using the developed algorithm and bench mark against results obtained from manual determination of the caller tone parameters. A standard caller tune file for Globacom was downloaded from the internet as ground truth caller tune. The file was converted with the format converter to a wav file and was fed into the developed algorithm to extract the call parameters for computation of necessary QoS. These parameters were then use as a standard against our dataset to compute the MSE as well as QoS estimation of MNPs base on the difference between the ground truth (downloaded tone) and our recorded tone for selected tone 1-10.

The normalized absolute error used in the measure of the performance of the system is given as:

$$E = \sqrt{(|a_{1m} - a_{1g}|^2)} / a_{1g} \tag{4}$$

Where a_{1m} is the measure parameter and a_{1g} is the graound truth. Analysis was carried out on the different hour ranging from less busy hour to the busiest hour in the dataset. Table 1 shows the caller tone parameters for Airtel dataset for day1. It is worthy of note that are all time used throughout this paper are measured in seconds.

Table 1: Caller tone parameters for Airtel dataset for day1.

S/N	Day 1 Monday-Time	SMT (sec) observe value	Inter Burst Time (sec)	Intra Burst Time (sec)	No of Burst
1	Airtel 2am T1	8.875	1.75	1.694	9
2	Airtel 2am T2	8.25	1.5	1.354	11
3	Airtel 2am T3	9.625	1.7	1.3625	10
4	Airtel 2am T4	9	1.725	1.338	10
5	Airtel 2am T5	10	1.675	1.338	10
6	Airtel 2am T6	10.25	1.613	1.475	10
7	Airtel 2am T7	9.875	1.625	1.341	11
8	Airtel 2am T8	10.25	1.488	1.663	10
9	Airtel 2am T9	8.954	1.681	1.417	10
10	Airtel 2amT10	8.603	1.472	1.638	10
11	Airtel 6am T1	7.375	1.5	1.329	11
12	Airtel 6am T2	6.75	0	0.625	1
13	Airtel 6am T3	5.125	0	11.875	1
14	Airtel 6am T4	5	0	11.25	1
15	Airtel 6am T5	4.875	0	6.25	1
16	Airtel 6am T6	6.375	0	1.125	1
17	Airtel 6am T7	4.75	1.75	1.35	11
18	Airtel 6am T8	5.625	1.5	1.364	11
19	Airtel 10am T1	9	1.5	1.352	11
20	Airtel 10am T2	12.5	1.725	1.313	11
21	Airtel 10am T3	7.625	1.688	1.363	10
22	Airtel 10am T4	17.75	1.738	1.375	10
23	Airtel 10am T5	18.5	1.523	1.295	11
24	Airtel 10am T6	25.875	0	0	0
25	Airtel 10am T7	8.5	2.075	1.35	10
26	Airtel 10am T8	7.375	1.463	1.638	10
27	Airtel 10am T9	9	3.818	1.341	11
28	Airtel 10am T10	18.75	1.625	1.364	11
29	Airtel 6pm T1	4.25	1.597	1.675	10

30	Airtel 6pm T2	10	1.488	1.363	10
31	Airtel 6pm T3	18.25	1.5	1.329	11
32	Airtel 6pm T4	5.5	1.725	1.363	10
33	Airtel 6pm T5	6.875	1.523	1.375	11
34	Airtel 6pm T6	10.125	1.638	1.375	11
35	Airtel 6pm T7	11.625	1.7	1.35	10
36	Airtel 6pm T8	9.125	1.511	1.364	11
37	Airtel 10pm T1	8.875	1.466	1.352	11
38	Airtel 10pm T2	11.5	1.75	1.338	10
39	Airtel 10pm T3	7.625	1.534	1.352	11
40	Airtel 10pm T4	9.75	1.6	1.352	11
41	Airtel 10pm T5	7.5	1.625	1.307	11
42	Airtel 10pm T6	13.25	1.488	1.35	10
43	Airtel 10pm T7	10.625	1.681	1.431	9
44	Airtel 10pm T8	8.625	1.625	1.375	11
45	Airtel 10pm T9	10.25	1.488	1.284	11
46	Airtel10pm T10	8.375	1.713	1.362	10

Performance Analysis of the technique in detecting Silent Mode Time (SMT)

Monday is regarded as the busiest day in a week in Nigeria. From observation of Globacom network, with 2am being regarded as the time with less traffic as most mobile users are sleeping. Performance analysis was carried out on the dataset obtained on Airtel network at 2am dataset. Performance analysis of the developed algorithm in detecting SMT from dataset is as shown in Table 1a and Table 1b.

Table 1a: Performance analysis of the Developed algorithm in detecting SMT at 2am on Monday for Airtel (Less Busy Hour).

Airtel at 2am	SMT from Design (sec)	Manual/Visual Determination of SMT (sec)	Error	% Error
Tone-1	8.875	8.953	0.0087	0.87
Tone-2	8.25	8.169	0.0099	0.99
Tone-3	9.625	9.725	0.0103	1.03
Tone-4	9	9.034	0.0038	0.38
Tone-5	10	10.006	0.0006	0.06
Tone-6	10.25	10.295	0.0044	0.44
Tone-7	9.875	9.904	0.0029	0.29
Tone-8	10.25	10.236	0.0014	0.14
Tone-9	8.954	8.76	0.0221	2.21
Tone-10	8.603	8.626	0.0027	0.27

Table 1b: Performance analysis of the Developed algorithm in detecting SMT at 6pm on Monday for Airtel (Busy Hour)

Airtel at 6pm	SMT from Design	Manual/Visual Determination of SMT	Error	% Error
Tone-1	4.25	4.187	0.015	1.5
Tone-2	10	7.531	0.3278	32.78
Tone-3	18.25	18.066	0.0102	1.02
Tone-4	5.5	5.586	0.0154	1.54
Tone-5	6.875	6.884	0.0013	0.13
Tone-6	10.125	9.997	0.0128	1.28
Tone-7	11.625	11.63	0.0004	0.04
Tone-8	9.125	9.238	0.0122	1.22

Table 1c: Performance analysis of the Developed algorithm in detecting SMT at 2am on Wednesday for Airtel (Less Busy Hour).

Airtel at 2am	SMT from Design	Manual/Visual Determination of SMT	Error	% Error
Tone-1	7	7.06	0.0085	0.85
Tone-2	8.5	8.549	0.00573	0.57
Tone-3	28	27.965	0.00125	0.13
Tone-4	7.62	7.679	0.00768	0.77
Tone-5	19.25	21.23	0.09326	9.33
Tone-6	21.375	21.427	0.00243	0.24
Tone-7	8.375	8.426	0.00605	0.61
Tone-8	5.875	5.859	0.00273	0.27

Table 1d: Performance analysis of the developed algorithm in detecting SMT at 6pm on Wednesday for Airtel (Busy Hour).

Airtel at 6pm	SMT(sec) from Design	Manual/Visual Determination of SMT(sec)	Error	% Error
Tone-1	9	8.98	0.00223	0.223
Tone-2	9.75	9.838	0.00894	0.894
Tone-3	8.375	8.396	0.0025	0.25
Tone-4	7.625	7.679	0.00703	0.703
Tone-5	9.75	9.846	0.00975	0.975
Tone-6	9.375	9.232	0.01549	1.549
Tone-7	8	7.979	0.00263	0.263
Tone-8	11.125	8.606	0.2927	29.27
Tone-9	7.875	7.968	0.01167	1.167

Table 1a and Table 1b shows some selected results obtained from the determination of SMT on a less-busy and most-busy hour on Monday while Table 1c and Table 1d shows the performance analysis of the developed algorithm in detecting SMT in less busy hour and busy hour respevtively. The result shows that the maximum error recorded from the developed algorithm as compared with the manually computed results is 4.21%. In some instances, the computed error was as low as 0.0% and the average percentage error was 1.2%. Also, it is evident in Table1a and Table that the error during the off peak i.e., 2am is far lesser than that of when the system is busy i.e., 6pm. This shows that as the system becomes busy, the QoS deteriorate. Similarly comparing, results of similar hours on Monday with that of Wednesday shows that more calls happened during the early hours of Wednesday than on Monday, thus the reason for increase in the error recorded on Wednesday. Generally, the results obtained from Table 1a-Table 1d shows that the developed algorithm is highly accurate in the determination of SMT and is a good measure of QoS parameters.

Performance analysis of the technique in detecting number of burst

The number of bursts in a caller tone plays a significant role in the determination of GSM QoS; hence the performance of the developed algorithm in accurate determination of the number of bursts is presented in this subsection. The performance has been done on 2am and 6pm dataset that represent both less busy and most-busy period on a busy day. Results obtained are discussed herewith.

Table 2a: Performance analysis of the developed algorithm in detecting Number of Burst at 2am on Monday for Airtel (Less Busy Hour).

Airtel at 2am	Number of Burst from Design	Manual/Visual Determination of Number of Burst	Error	% Error
Tone-1	10	11	0.0909	9.09
Tone-2	11	11	0	0
Tone-3	10	10	0	0
Tone-4	10	10	0	0
Tone-5	10	10	0	0
Tone-6	10	10	0	0
Tone-7	11	11	0	0
Tone-8	10	11	0.0909	9.09
Tone-9	10	10	0	0
Tone-10	10	10	0	0

Table 2b: Performance analysis of the Developed algorithm in detecting Number of Burst at 6pm on Monday for Airtel (Less Busy Hour).

Airtel at 6pm	Number of Burst from Design	Manual/Visual Determination of Number of Burst	Error	% Error
Tone-1	10	11	0.0909	9.09
Tone-2	10	11	0.0909	9.09
Tone-3	11	11	0	0
Tone-4	10	11	0.0909	9.09
Tone-5	11	11	0	0
Tone-6	11	11	0	0
Tone-7	10	11	0.0909	9.09
Tone-8	11	11	0	0

Table 2c: Performance analysis of the Developed algorithm in detecting Number of Burst at 2am on Wednesday for Airtel (Less Busy Hour).

Airtel at 2am	Number of Burst from Design	Manual/Visual Determination of Number of Burst	Error	% Error
Tone-1	10	10	0	0
Tone-2	11	11	0	0
Tone-3	11	11	0	0
Tone-4	11	10	0.1	10
Tone-5	11	10	0.1	10
Tone-6	11	11	0	0
Tone-7	10	10	0	0
Tone-8	10	10	0	0

Table 2d: Performance analysis of the Developed algorithm in detecting Number of Burst at 6pm on Wednesday for Airtel (Less Busy Hour).

Airtel at 6pm	Number of Burst from Design	Manual/Visual Determination of Number of Burst	Error	% Error
Tone-1	11	11	0	0
Tone-2	10	10	0	0
Tone-3	11	10	0.1	10
Tone-4	11	10	0.1	10
Tone-5	11	11	0	0
Tone-6	11	11	0	0
Tone-7	11	10	0.1	10
Tone-8	10	11	0.09091	9.091
Tone-9	11	11	0	0

Table 2a and Table 2b shows the results obtained in accurately detecting number of bursts in different caller tone on a Monday while Table 2c and Table 2d shows the results obtained in accurately detecting number of bursts in different caller tone on a Wednesday. Similar to the analysis in detecting SM, two scenarios have been reported in this subsection also. It can be observed that the error in the determination of number of bursts shows significant difference between 2am calls and 6pm calls. This also justify the earlier assertion that MNP recorded more calls during 6pm peak duration. Furthermore, comparing Monday data to that of Wednesdays shows significant increase in error; does an indication of reduced QoS on those days. In conclusion, the developed algorithm has the capability to accurately determine the number of bursts in different scenarios.

Performance analysis of the technique in detecting caller tune frequency

Figure 8a shows a typical result obtained in the determination of the Caller Tune frequency both in time domain and in Frequency domain. The FFT equation given in equation 4 was used in the determination of the caller tone. Figure8a-e shows the results obtained in determining the caller tune frequency from the recorded signal.

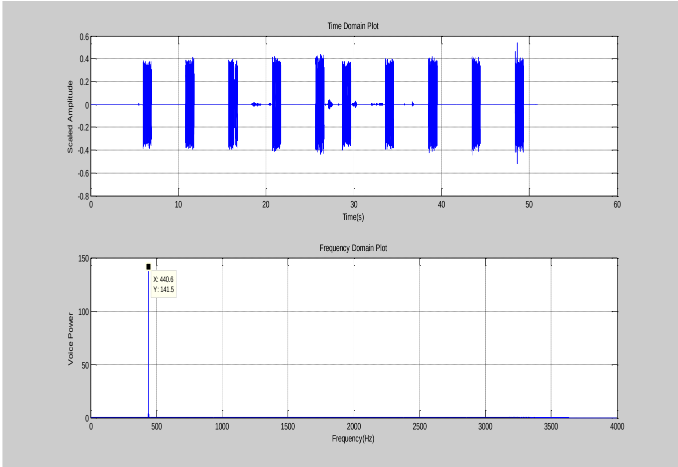

Figure 8a: Time and Frequency domain plot of the caller tune.

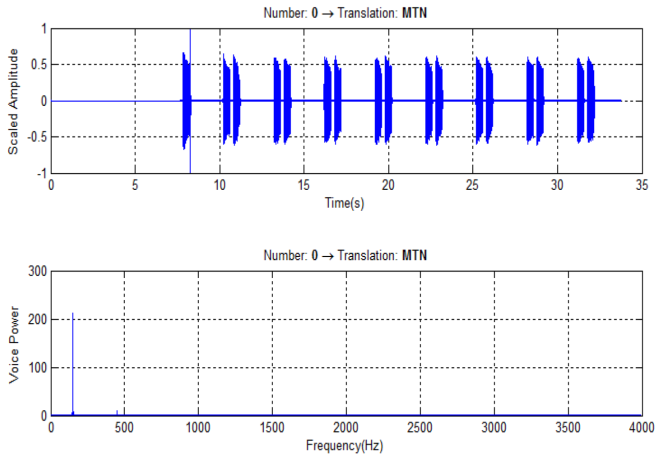

Figure 8b: Time and Frequency domain plot of MTN caller tune.

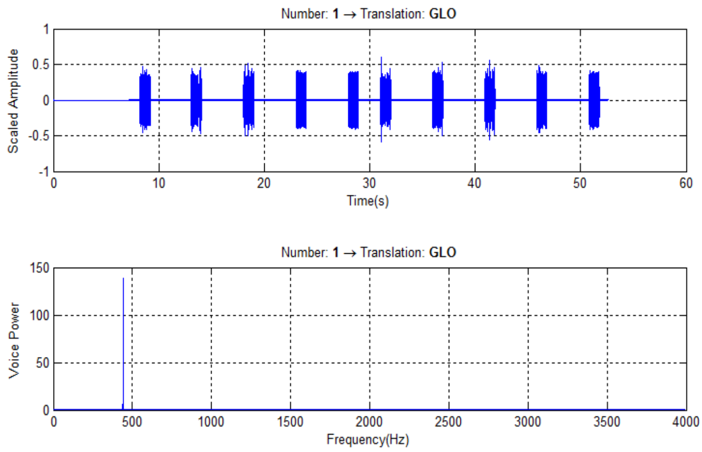

Figure 8c: Time and Frequency domain plot of GLO caller tune.

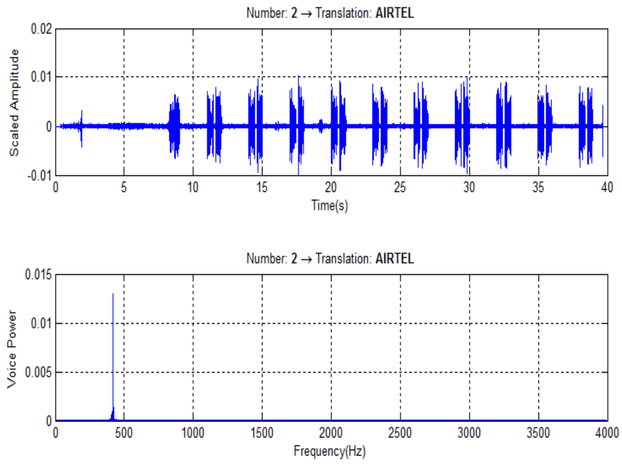

Figure 8d: Time and Frequency domain plot of Airtel caller tune.

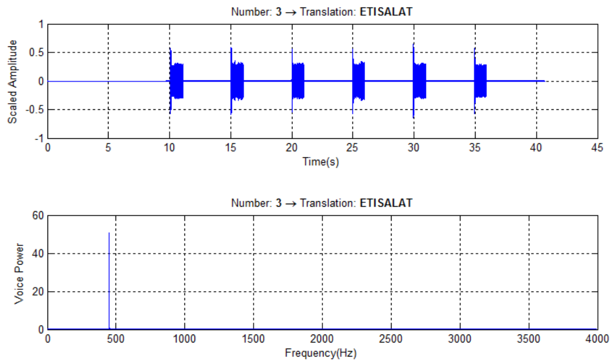

Figure 8e: Time and Frequency domain plot of Airtel caller tune.

Figure 8b to Figure 8e shows a typical result obtained in the determination of the caller tune frequency both in time domain and in frequency domain. Their frequencies are about 150Hz, 440Hz, 420Hz and 450Hz for MTN, GLO, Airtel and Etisalat caller tune respectively.

Performance analysis: QoS estimation

In estimating the Qos of a caller tune, the Euclidean Distance Accuracy Measure have been used. The Euclidean distance measures the closeness between the determined caller tune parameters and the ground truth. Parameters used to compute the final MSE include: SMT; InterBurst Time; IntraBurst Time; No of Burst; Call Duration and call Frequency. The MSE is computed as follows:

The Euclidean distance accuracy measure (Υ) is given as

$$\Upsilon = \sqrt{|x_{1R} - x_{1G}|^2 + |x_{2R} - x_{2G}|^2 + \ldots + |x_{5R} - x_{5G}|^2} \tag{5}$$

Where x_{1R} is the SMT of the recorded caller tune under test; x_{1G} is the SMT of the ground truth caller tune; x_{2R} is the InterBurst Time of the recorded caller tune under test; x_{2G} is the InterBurst Time of the ground truth caller tune; x_{3R} is the IntraBurst Time of the recorded caller tune under test; x_{3G} is the IntraBurst Time of the ground truth caller tune; x_{4R} is the No of Burst of the recorded caller tune under test; x_{4G} is the No of Burst of the ground truth caller tune and x_{5R} is the Call Duration of the recorded caller tune under test; x_{5G} is the Call Duration of the ground truth caller tune. The rating of the QoS is as given in Table 3.

Table 3: QoS Rating.

ϒ	QoS Status
0<= ϒ <=0.9999	Good Qos
1.0000<= ϒ <=1.9999	Averagely Good
2.0000<= ϒ <=2.9999	Poor
ϒ >=3.0000	Extremely Poor

Table 4 shows the analysis on the dataset for different MNPS at 2am. It can be observed from the Table 4 that mobile network connections between Airtel network and GLO network at 2am is good as the average MSE is 0.78 which is rated high with respect.

Table 4: QoS estimation using the developed technique.

Airtel 2am for 3 consecutive Mondays	SMT	Inter Burst Time (sec)	Intra Burst Time (sec)	No of Burst	Call Duration (sec)	Error	MSE	QoS
Ground Truth	8.75	1.575	1.375	10	38.36			
Tone-1	8.875	1.738	1.363	10	39.99	0.169	0.411092727	GOOD Qos
Tone-2	8.25	1.5	1.352	11	39.69	0.32293	0.566827269	GOOD Qos
Tone-3	9.625	1.7	1.363	10	40.37	0.41618	0.645118566	GOOD Qos
Tone-4	9	1.725	1.338	10	39.7	0.11458	0.338495676	GOOD Qos
Tone-5	10	1.675	1.388	10	40.7	0.20265	0.450171904	GOOD Qos
Tone-6	10.25	1.613	1.475	10	41.23	0.13772	0.371102619	GOOD Qos
Tone-7	9.875	1.625	1.341	11	40.99	0.24069	0.490605117	GOOD Qos
Tone-8	10.25	1.488	1.663	10	41.8	0.47307	0.687802078	GOOD Qos
Tone-9	8.954	1.681	1.417	10	39.21	0.46603	0.682664254	GOOD Qos
Tone-10	8.603	1.472	1.638	10	39.3	0.32179	0.567265148	GOOD Qos
Tone-1	5.25	1.813	1.363	10	37	0.84782	0.920769902	GOOD Qos
Tone-2	10.25	2.432	1.318	11	51.54	1.81979	1.348996808	AVE. GOOD Qos
Tone-3	8.125	1.513	1.67	11	41.7	1.04319	1.021364948	AVE. GOOD Qos
Tone-4	10	2.898	1.1917	12	64.96	1.94278	1.39383505	AVE. GOOD Qos
Tone-5	7.625	1.409	1.432	11	38.96	1.48788	1.21978758	AVE.GOOD Qos
Tone-6	8.375	1.725	1.375	10	34.46	0.56885	0.75422134	GOOD
Tone-7	4.5	2.375	1.51	11	48.81	1.4541	1.205862687	AVE. GOOD Qos
Tone-8	7.125	3.925	1.738	10	63.76	1.78416	1.335723428	AVE. GOOD Qos
Tone-9	8.375	1.5	1.663	10	40.4	1.2028	1.096722364	AVE. GOOD Qos
Tone-1	8.875	1.5	1.364	11	40.43	0.34024	0.583300623	GOOD Qos
Tone-2	8.875	1.511	1.375	11	40.69	0.02183	0.147745449	GOOD Qos
Tone-3	9.625	1.523	1.341	11	41.19	0.12946	0.35981121	GOOD Qos

Tone-4	9.375	1.511	1.318	11	40.51	0.06751	0.25983351	GOOD Qos
Tone-5	15.5	1.45	1.413	10	44.22	0.94827	0.973793753	GOOD Qos
Tone-6	8.5	1.489	1.341	11	39.74	0.73078	0.854854662	GOOD Qos
Tone-7	8.875	1.489	1.364	11	40.31	0.07561	0.274976831	GOOD Qos
Tone-8	11.75	4.556	1.347	9	64.92	3.18852	1.785641433	AVE. GOOD Qos
Tone-9	9.375	3.795	1.329	11	65.8	0.6183	0.786320889	GOOD Qos
Tone-10	11.125	4.556	2.347	9	64.3	1.3578	1.165245758	AVE. GOOD Qos

CONCLUSION

This study was undertaken to determine call setup time and ringtone quality in GSM network independence of the mobile network provider. For the purpose of this study GLOBACOM Nigeria was used as the source. We gathered ringtone every four hours for a month in one of the sub-urban regions of Abuja using a SAMSUNG GALAXY S4 and SAMSUNG GALAXY A3 although data were collected from other networks too to have a general knowledge of the ring pattern. Our data revealed that ringtone parameters for the networks differs even when measurement is taken in the same region and also, we discover that SMT has a great impact on the quality of service of the mobile network, the longer the SMT the higher the possibilities of call drop hence the lower the QoS of the mobile provider. Mean square error (MSE) is used as a performance index that indicates how well the algorithm can adapt to a given solution. A small minimum MSE is an indication that the QoS estimator design is accurately modelled and can predict, adapt and converge to a given solution. However, a very large MSE usually indicates that the system is not accurately modelled.

REFERENCES

[1] B. V. Sireesha and S. Varadarajan, "Increasing Of Call Success Rate In GSM Service Area Using RF Optimization," vol. 1, no. 4, pp. 1479–1485.

[2] Z. A. Dahi, E. Alba, and G. Luque. "A takeover time-driven adaptive evolutionary algorithm for mobile user tracking in pre-5G cellular networks". Applied Soft Computing. 2021 Oct 27:107992.

[3] H. Hassan, R. W. Goshie, A. Jimada, R. Oyewumi, S. Amata. "Customers' Intension to Change Mobile Communication Networks Regarding Quality of Information Services Provision in Zaria Metropolis". i-manager's Journal on Mobile Applications and Technologies. 2019;6(1):27.

[4] Y. H. Lu, C. Y. Li, Y. Y. Li, S. H. Hsiao, T. Xie, G. H. Tu, and W. X. Chen. "Ghost calls from operational 4G call systems: IMS vulnerability, call DoS attack, and countermeasure". In Proceedings of the 26th Annual International Conference on Mobile Computing and Networking 2020 Apr 16 (pp. 1-14).

[5] A. M. Aibinu, A. J. Onumanyi, and S. Africa, "Automatic determination of call setup time and and ring tone quality of service in GSM Network" International Engineering Conference IEC, September, 2015 (pp.424-430).

[6] E. Amaldi, A. Capone, and F. Malucelli, "Radio planning and coverage optimization of 3G cellular networks," *Wirel. Networks*, vol. 14, no. 4, pp. 435–447, 2008, doi: 10.1007/s11276-006-0729-3.

[7] F. C. De Gouveia and T. Magedanz, "quality of service in telecommunication networks." Telecommunication systems and technologies, 2(5), pp. 77-84

[8] M. Alipio, N. M. Tiglao, F. Bokhari, and S. Khalid. "TCP incast solutions in data center networks: A classification and survey". Journal of Network and Computer Applications. 2019 Nov 15;146:102421.

[9] K. N. R. S. V. Prasad, H. K. Rath, and A. Simha, "Wireless mobile network planning and optimization." In *2014 sixth International Conference on Communication Systems and Networks (COMSNETS)* (pp. 1-4). IEEE

[10] J. Agajo, A. L. Theophilus, V. E. Idigo, K. I. Apkado, F. Polytechnic, and E. State, "Optimization of Network Performance in Wireless Communication Networks." The Pacific Journal of Science and Technology. 2012 vol.13(1), (pp.334-350)

[11] N. Khan, M. U. Akram, A. Shah, N. S. Alghamdi, and S. A. Khan. "Capturing the real customer experience based on the parameters in the call detail records". Multimedia Tools and Applications. 2021 Jun 4:1-23.

[12] S. R. Srivastava, S. Dube, G. Shrivastaya, and K. Sharma. "Smartphone triggered security challenges—Issues, case studies and prevention." Cyber Security in Parallel and Distributed Computing: Concepts, Techniques, Applications and Case Studies. 2019 (pp.187-206).

[13] M. Kollár, "Evaluation of Real Call Set Up Success Rate in Gsm," vol. 8, no. 3, pp. 53–56, 2008.

[14] B. Haider, M. Zafrullah, and M. K. Islam, "Radio Frequency Optimization & QoS Evaluation in Operational GSM Network," *Lect. Notes Eng. Comput. Sci.*, vol. 2178, no. 1, pp. 393–398, 2009.

[15] M. E. Hossain, M. W. Rahman, M. T. Islam, and M. S. Hossain. "Manifesting a mobile application on safety which ascertains women salus in Bangladesh". International Journal of Electrical and Computer Engineering. 2019 Oct 1;9(5):4355.

[16] B. Appiah, A. Yoshikawa, L. Asamoah-Akuoko, D. A. Anum, I. A. Kretchy, E. Samman, A. Goulart, J. Nwokike, A. N. Dodoo, D. M. Darko, and A. Rene. "Consumer reporting of suspected adverse drug reactions: modelling the acceptance of mobile phone caller tunes to raise awareness". Journal of Pharmaceutical Health Services Research. 2020 Mar;11(1):39-48.

[17] D. Aliu and M. O. Momoh (2021) "Optimal Resource Scheduling Algorithm for OFDMA-based Multicast Traffic Delivery over WIMAX Networks using Particle Swarm Optimization" International Journal of Software Engineering & Computer Systems (IJSECS). VOL.7, (1), August 2021. 50-63

[18] R. Budhiraja and J. S. Jadon, "Study And Implementation Of Drive Test For Development Of GSM Network," *Int. J. Eng. Trends Technol. – Vol. 4 Issue 10 - Oct 2013*, vol. 4, no. 10, pp. 4352–4357, 2013.

[19] O. R. Abubakre, O. Ubadike, A. M. Aibinu, T. A. Folorunso, and M. O. Momoh, (2021). "Artificial Neural Network Embedded Optimal Mobile Communication System". *Covenant Journal of Informatics and Communication Technology*, *9*(2).