Improving Quality of Service in GSM By Reducing Probability of Call Blocking Through Network Dimensioning Using Erlang B Model and Congestion Control Algorithm

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ABSTRACT : Call blocking probability is a key performance metric for any telecommunication protocol. Unexpectedly, the level of patronage being experienced in Global System for Mobile Communication (GSM) in Nigeria is overwhelming. This is as a result of freedom of calling from anywhere at any time and clarity of the voice enjoyed in GSM since it is on a digital technology platform. This has brought a lot of congestion in the network resulting in poor services by the operators. This research has developed a control algorithm for the management of the congestion experienced in the GSM network in Nigeria. It explores the use of Erlang-B in determining the appropriate probability level for some range of subscribers. Thereafter, when there is congestion, block time sharing, dynamic allocation without time slicing, dynamic allocation time slicing with signal sensing, frequently recent call allocation, and priority allocation algorithms were developed to manage the congestion. Furthermore, a hybrid algorithm was developed that integrates all the algorithms together in other to manage the congestion considering all the strengths and constraints of each algorithm. If the recommended congestion management algorithm is followed comprehensibly, the congestion problem on the GSM network will be reduced drastically which in turn reduces call block. Also, from the research conducted by NCC and others in quest for ameliorating the call block problem, it was noted that from the average number of subscribers that dialed up to three or more times before getting connected, 58% of call blocking rate was recorded which after using more number of timeslots (which depicted more available channels) in this work was reduced as showed in the simulation results and will also be reduced the more using the developed congestion control algorithm technique. The simulation codes and program are shown on the appendix in chapter five.

I. INTRODUCTION

1.1 BACKGROUND OF THE STUDY

From time immemorial, information and communication have formed the basis of human existence. Before the advent of GSM in Nigeria, telephones were luxury that only a few privileged Nigerians enjoyed.
However, with the advent of GSM in Nigeria in August 2001, mobile telephony has rapidly become the most popular method of voice communication in the country. Its growth has been so rapid that Nigeria has been rightly described in various media as one of the fastest growing GSM market in the world.
The first generation of cellular telephone system, which was analog system, launched in 1960s had the limitations of incompatibility among the various analog system available amongst other limitations which led to the invention of second generation digital system called GSM with a wide spectrum allocation.
Following the rollout of GSM services across the nation, the socio-economic landscape of Nigeria has been positively altered. Its explosive growth has brought huge revenues to the operators as well as the government through tax and license fees.
Similarly, the citizenries have benefited immensely from the services not only as a means of communication, but it has provided job opportunities for thousands of people.
However, the principal development that mars these benefits is the aggressive complaining raised by GSM subscribers regarding to the abysmal quality of service (Qos) rendered by the GSM operators in the country. The unfortunate aspect of this evil is the fact that all the GSM subscribers irrespective of the operator are being affected. Based on this ugly experience, this research study was focused on the causes of the problem and ways of ameliorating the observed defects. Generally, there are five frequency bands designated by international Telecommunications Union (ITU) for the operation of the GSM mobile phone: **GSM-400, GSM-850, GSM-900, GSM-1800 and GSM-1900.**

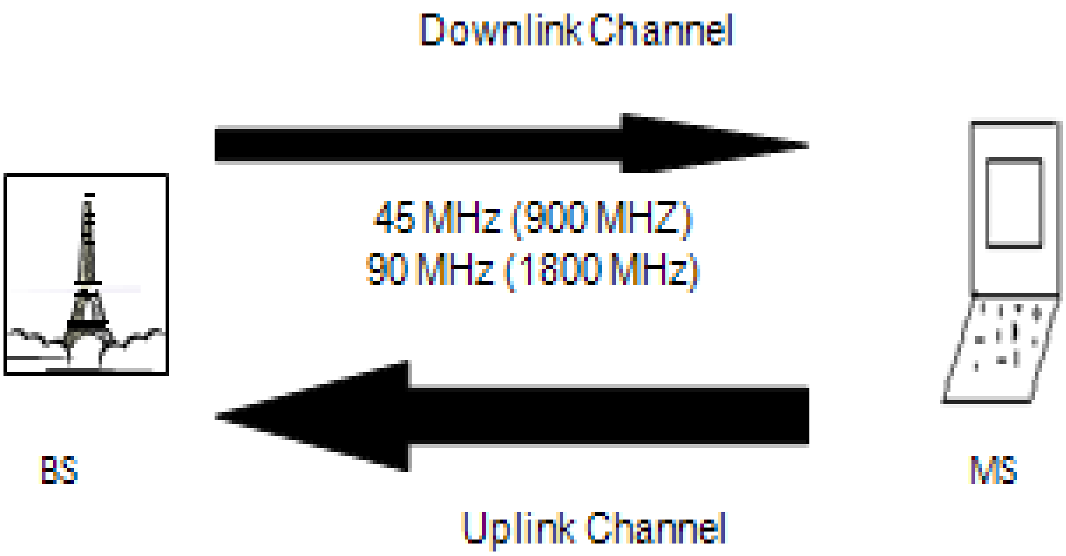
In Nigeria, **GSM-900 and GSM-1800** are being used. Here, **GSM-1800** uses **1710-1785 MHZ** to send information from the mobile station to the base transceiver station (uplink) and **1805-1850MHZ** for the other direction (downlink). Also, **GSM-900** uses **890-915 MHZ** for the forward or uplink direction while the reverse or downlink direction is on **935-960 MHZ** full details on other GSM frequency bands using across the world is shown in table 1 below.

Table 1.1 World-Wide GSM frequency B and

System	**B and (MHZ)**	**Uplink (MHZ)**	**Downlink (MHZ)**
GSM-400	450	4504-4576	460.4-4467
GSM-400	480	478.5-486.0	488.8-496.0
GSM-850	850	824.0-849.0	869.0-894.0
GSM-900	900	890.0-915.0	935.0-960.0
GSM-900(£.GSM)	900	880.0-915.0	925.0-960.
GSM-900(R-GSM)	900	876.0-880.0	921.0-925-0
GSM-1800	1800	1710.0-1785.0	1805.1880.0
PCS-1900	1900	1850.01910.0	19300-19900

The method of sharing the radio spectrum bandwidth ofGSM is known as Time- and Frequency-Division Multiple Accesses (TDMA and FDMA). FDMA part involves the division by frequency of the maximum 25 MHZ bandwidth into 124 carrier frequencies spaced 200 KHZ apart (Harte et al, 1999). The actual number of carrier in GSM is 125 but due to interference to other systems, the very first carrier is not used thus reduced the carrier to 124. One or more carrier frequencies are allocated to each base station. Each of these carrier frequencies is then divided in time, using TDMA scheme into 8 timeslots as seeing in fig 2.0. Each of these timeslots is a physical channel occupied by an individual user (carries control and a traffic data in a burst form) (Mehrotra, 1997). Typical GSM handsets carrier channel, thus allows up to eight users to simultaneously share a single radio channel.

Figure 1. GSM Duplex Radio (Source: Mehrotra, 1997)



The management of this radio frequency among subscribers remains a problem all over the world including Nigeria. A lot of literatures have been written in respect of this problem (Gupta and Sachan, 2007; Candan and Salamah, 2006). The congestion experienced today is a result of this problem at hand. This research took a critical look at the GSM network in Nigeria and come up with algorithms that can be adapted on the GSM network to minimize this congestion.

However, the data from the present study is compared with the results of the previous studies in order to ascertain the accuracy of the work.

1.2 STATEMENT OF PROBLEM

With the poor and epileptic quality of services presently rendered by GSM operator in Nigeria, it is possible that in no distance time, a GSM subscriber of a particular network operator may not be able to get or established a communication link with another subscribers from another network if nothing is done about it.

1.3 OBJECTIVES OF THE STUDY

This research work had some of its objectives to include:

- **To find the causes of call blocks**
- **To analyse network congestion and its causes relating to call blocking**
- **To obtain rate of setting up a call**
- **To find the causes of unsuccessful call handovers Relating to call blocking**
- **To obtain call completion success rate.**
- **To determine the appropriate probability level for range of network subscribers using Erlang B model.**
- **To develop an Algorithm for management of network congestion.**

1.4 SIGNIFICANCE OF THE STUDY

It is clear and obvious that quality of service is an important key performance indicator (KPI) that is used in determining the efficiency of an industry in terms of services rendered, so that of the telecommunication industry cannot be left out. And for the consumers in the industry, it is expected that maximum satisfaction be derived from any services paid for, therefore, the activities of the GSM operators in the country should always be monitored by government and the GSM operators themselves should wakeup in the areas of quality of signal. This problem, if ignored, might increase the difficulty in inter-Network connectivity, affect the social security of the country negatively and break the minimum performance levels jointly agreed between GSM operators in Nigeria, consumer representatives and NCC.

1.5 SCOPE AND LIMITATION OF STUDY

To be sincere, a lot of issues and problems were encountered in the process of information gathering during this research work which include, disruption of movement as a result of bad road, withholding of information by the relevant agencies amongst other factors too. Also this research work mainly focused on call block problems and its ways of reduction as a key player in quality of service assessment in communication industries.

II. LITERATURE REVIEW

2.1 INTRODUCTION

Following the effort to find the solution to the poor quality of services provided by GSM operators in Nigeria, Nigeria Communication commission, the body responsible for regulation of GSM operation in Nigerian carried out a research in 2005 to find causes of the problems and ways of ameliorating it.

This research was centered on customer's complaints method through the use of questionnaire. The investigation was conducted for period of 4 months (May 2005-August 2005) in the country federal capital Territory (FCT) and some other selected cities in all the six geo-political zones (North-West, North-Central, North-East, South-West, South-South and South-East) of the country, where all of the three major GSM networks considered were operating. The GSM network studied are Celtel, MTN and GLO. The study was conducted using structured questionnaire. The primary data obtained from this investigation was later compared with secondary data from other previous related studies.

2.2 STRUCTURE OF THE QUESTIONNAIRE

This involved the designing and administering of the well structured research questionnaire to the customers of the three GSM networks in the studied areas. The questionnaire was divided into two sections.

The first section is the introductory part, where the aim of the study was stated. This section also contains words of assurance and encouragement, assuring respondents that the information provided by them would be treated with utmost confidentiality.

The second section of the questionnaire consists of questions related to objectives of the study. The section was divided into four subsections. The first subsection contains four questions on ***NETWORKACCESSIBILITY.*** Questions on call set up rate such as how easy is it to set up call, number of attempts they do make before having a successful call, etc were asked.

In the second subsection, questions on retainability on the network after successful call set up were asked. Question such as how often do they experience call termination before completing their conversation were asked. Two questions were asked in this subsection from which service retainability degree of each of the three networks studied were evaluated.
The third subsection contains three questions on the correction quality or service integrity of each of the networks considered. Questions on voice quality and SMS delivery of each operator were asked.
In the fourth subsection, two questions were asked on network coverage or network availability of the three GSM operators. In all, a total of eleven questions were asked, which were structured in such a way that one question leads to the next.

FIGURE 2.1 Call setup success Rate (Easy/Difficult)

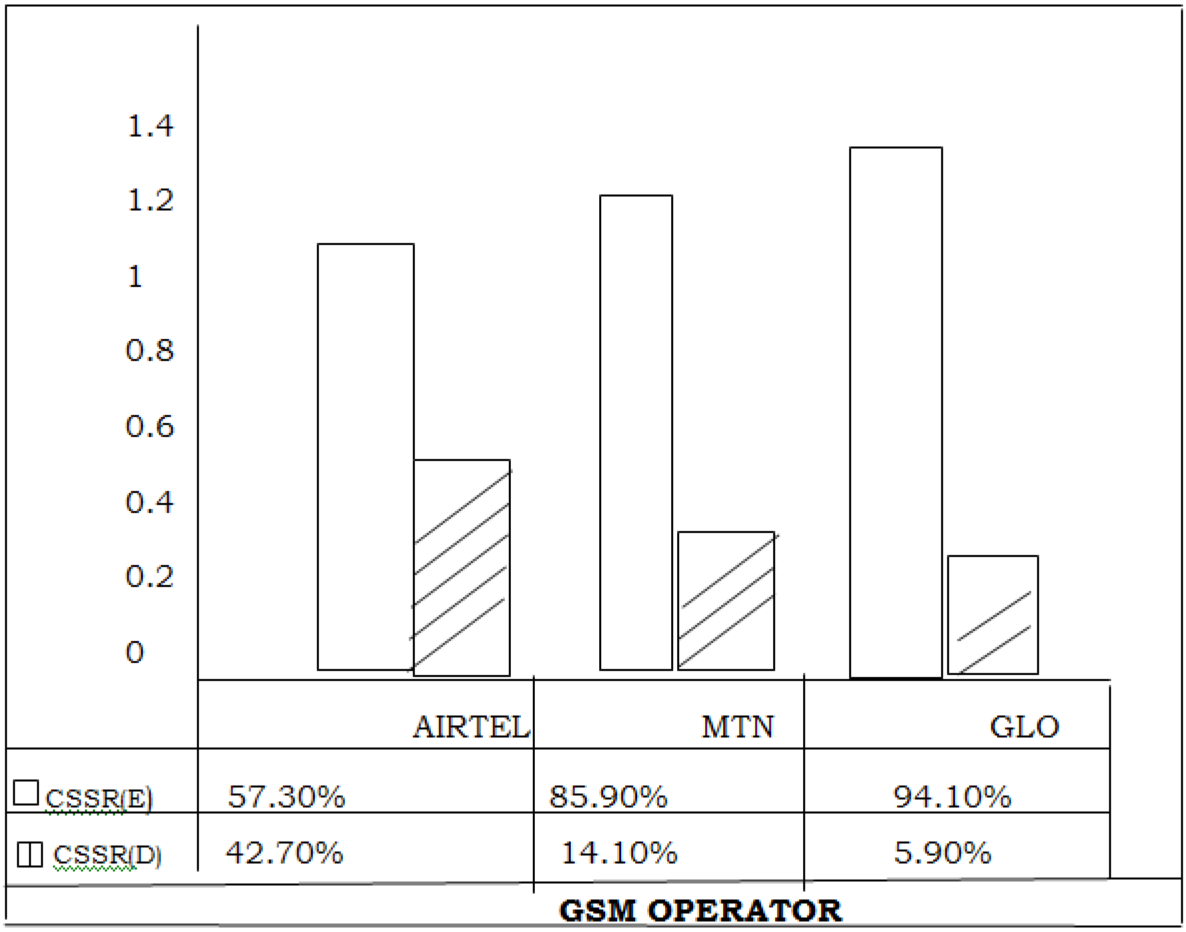
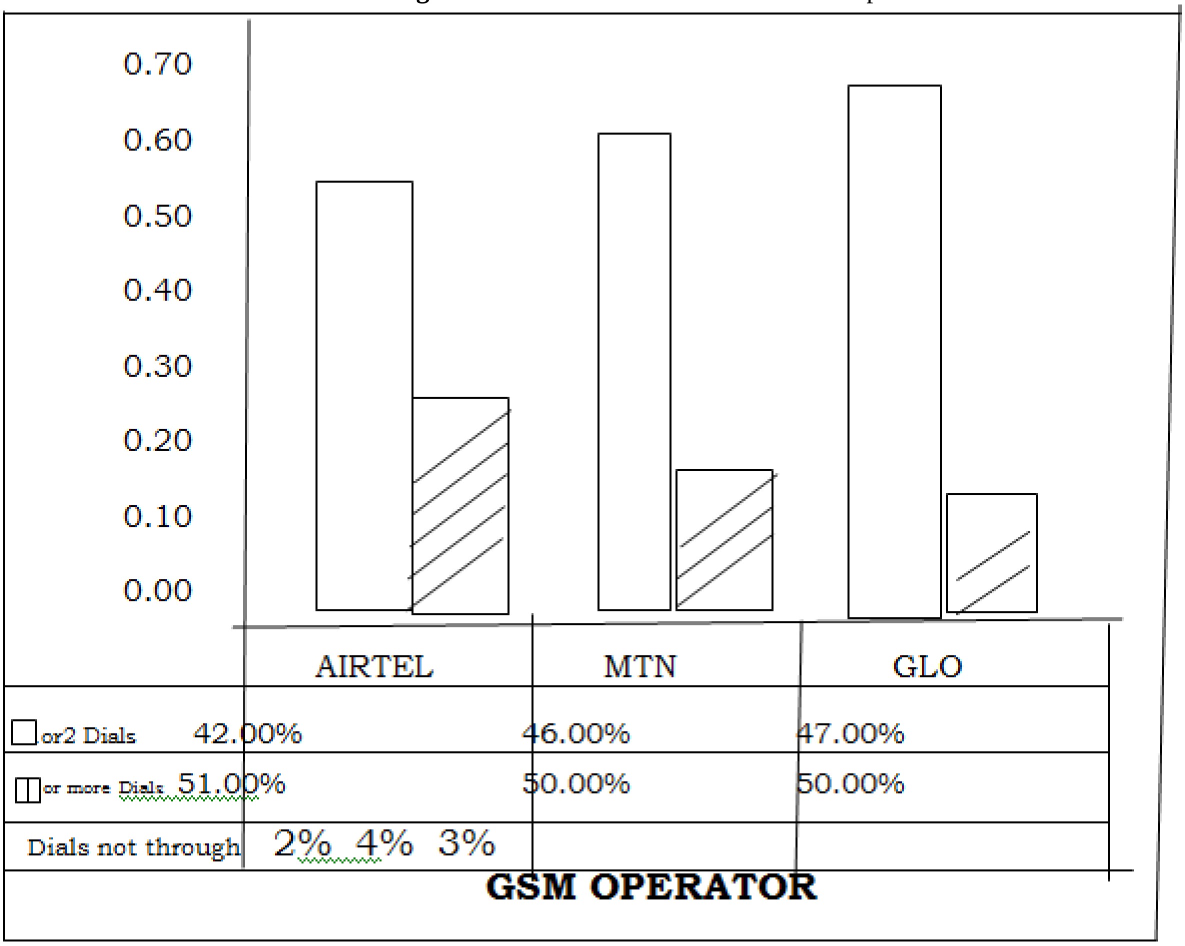


Figure 2.1 shows the rate of call setup or accessibility to each of GSM network. CSSR (E) means easy call setup rate while CSSR (D) means difficult call setup rate. The figure 2.1 shows that among the three GSM operators studied, Glo has the highest easy call setup success rate, while Airtel has the lowest easy call setup success rate. This implies that, the accessibility of MTN is easier than that of Airtel.

Figure 2.2: Number of dials before call setup.



The degree of accessibility of figure 2.2, which indicates the number of attempts a caller makes before a call can be established. From the figure it was found for every 100 calls made on the Airtel network, there is high probability of having 47 success calls with the first or second dials while 51% of those successful calls only occurred with three or more numbers of attempts on Airtel networks. The figure reveals that accessibility to both Glo and MTN network can be accomplish with fewer number of attempt.

2.3 ADMINISTRATION OF THE QUESTIONNAIRE

This phase of the work was carried out in the selected areas in the FCT and some cities in each of the six geo-political zones where the three GSM operators are operating.

Table I shows the detail on the retrieved questionnaires from each of the studied areas. The responses gathered were converted to percentage so that the result analysis can be on the equal basis.

Table 2.1 Questionnaire distribution

GSM NETWORK

ZONE	CELTEL	MTN	GLO
North-West	1060	1383	699
North-Central	559	11208	840
North-East	476	364	253
South-West	839	1074	534
South-South	832	1774	677
South-East	229	419	768
FCT Abuja	473	587	506
Total	4468	6807	4277

2.4 CALL SET UP FAILURE:

This an important parameter used in determining network accessibility. It is the ability of a subscriber to initiate a call and granted access. Technically, during a GSM call setup, a speech call is assigned from a SDCCH (Stand alone dedicated channel) to a TCH (traffic channel). If the TCH selected suffers from interference, then the assignment will fail. And assignment failure message will be sent to the MSC. The call will then be re-established back. According to the survey carried out by NCC, all the three major operators were found to perform poorly in the area of number of times that users dial before connection is made. The survey shows that only less than half of the subscribers on each of the networks do get their calls through on the first or second dial. ***(Airtel-49%, MTN-46%, GLO-47%).***

In other words, subscribers 3 times or more were (***Airtel-49%, GLO-50%, MTN-50%).*** Results of sampled opinion are as shown in table below.

Table 2.2 showing subscribers dialing 3 or more times before getting connected.

CITY	AIRTEL %	MTN %	GLO %
Abuja	56	63	67
Kaduna and Zaria	39	50	45
Maiduguri	61	53	39
Kano	41	42	53
Jos	33	38	48
Bauchi	57	68	49
Ibadan	46	41	42
Calabar	45	79	50
Port Harcourt	4	47	20
Owerri	63	54	45
Enugu	37	59	61
Benin	51	60	58
Lagos	35	39	50
Kwara	46	36	33

AGGREGATE, ***AIRTEL =47%, MTN = 50%, GLO = 49%***

Table 2.3: Showing subscribers connected to numbers not dialed.

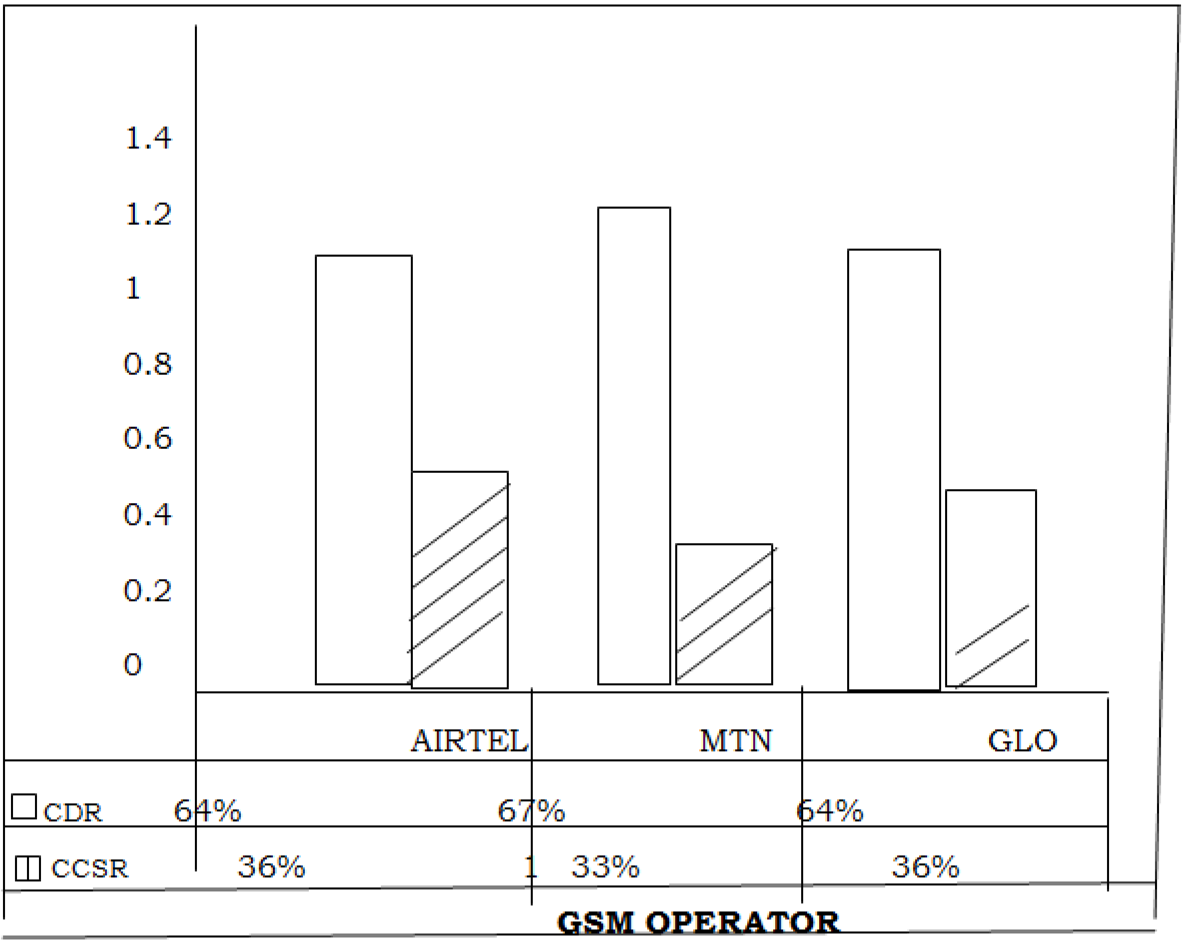
CITY	AIRTEL %	MTN %	GLO %
Abuja	67	60	60
Kaduna and Zaria	29	67	59
Kano	74	77	73
Maiduguri	47	34	34
Jos	57	64	74
Bauchi	32	36	58
Calabar	65	59	75
Owerri	52	62	51
Enugu	69	57	60
Benin	62	71	70
Port Harcourt	84	78	79

AGGREGATE, AIRTEL =64%, MTN = 67%, GLO = 64%

2.5 CALL RETENTION/CALL DROP:

Call retention is the ability to retain a GSM call after it has been established while dropped call is a situation where by an established call is abruptly terminated while conversation is ongoing. It is a common occurrence in Nigeria GSM system that communication is terminated unexpectedly while conversation is on-going.

Figure 2.3: **Call drop and call completion success rate.**



In all the networks considered, it was found that 36% is the maximum CCSR value the subscribers do experience. This implies that for every 100 successful call setup, only 36 of them will not drop before the parties completed there conversation. The value reveals that the retainability on all the three GSM network is low. This is an indication that their services are unreliable and unsatisfied

A survey conducted by NCC recently clearly showed that one of the most important customers perceived problems that affects quality of service is in the area of dropped calls. This is as depicted in table 2.4 below.

Table 2.4: Summary of Dropped calls and their locations

CITY	AIRTEL %	MTN %	GLO %
Abuja	67	69	60
Kaduna and Zaria	29	67	59
Kano	74	77	73
Maiduguri	47	34	34
Jos	57	64	74
Bauchi	32	36	58
Calabar	65	59	75
Owerri	52	62	51
Enugu	69	57	60
Benin	62	71	70
Port Harcourt	84	78	79

AGGREGATE, AIRTEL =64%, MTN = 67%, GLO = 64%

CONGESTION: Congestion is a phenomenon in telecommunication system that occurs when more subscribers attempt simultaneously to access the network than it is able to handle. This is a situation where subscriber's numbers has completely overgrown network capacity.

In quest to find a lasting solution to the poor quality of services rendered by GSM operator in Nigeria, some scholars and agencies have made some researches and suggestions which when applied could not solve the problem completely because of one or two factors that wasn't put into consideration.

Adegoke A.S and Babalola I.T of the department of Electrical Engineering, Yaba College of Technology, in 2008, conducted a research on quality of service assessment.
In their, work they made use of ***NETWORK STATISTICS*** as the mechanism to monitor, analyze and evaluate quality of service.

Here, hypothesis tests were considered to compare network operator's performance. For this purpose, two different hypothesis testing methods were conducted. These were based on classical statistical approaches, namely ***chi-square (X^2)*** and ***fisher's exact test***. By this way, firstly contingency tables were obtained. Then, by using chi-square distribution, hypothesis tests were derived to compare network operator's performance.
Lastly, hypothesis test were repeated using fishers exact test which is a reliable method if the sample sizes are small.

2.6 CHI-SQUARE HYPOTHESIS TESTING METHOD

Chi-square distribution is one of the most widely used probability distribution in inferential statistics, e.g in hypothesis testing, or in construction of confidence intervals. The chi-square distribution is the distribution of a sum of the square of K independent standard normal random variables. In many problems, it is not assumed that the available observation come from a particular parametric family of distributions. The chi-square test is widely used in the cases where no special assumptions are made about the form of the distribution. In chi-square test conducted for speech comparison of networks, contingency tables were generated to examine dependency of the networks. It is known that a contingency table, as it is typically represented by a table having R rows and C columns(R X C), is a tabular arrangement of count data representing how much row factor are related to the column factor.

In this study, samples from the three networks are selected at randomly. Efforts focuses on assessing paired observations (number of good speech sample /number of samples), and the reports the results in a 2X2 contingency table that are independent of each, as discusses in several works in the literature.

In other to define number of good sample, a threshold value is needed. This value indicates the minimum quality of speech perceived by users (PESQ) value that represents the acceptable speech quality. Thus, for set of speech samples, each sample is compared with the threshold and then good samples among the total number of sample can be obtained. Three different contingency tables were constructed since the tests are performed two by two, pair of the networks or operators. That is, operator A is compared with B and then operator C separately, and finally operator B is compared with operator C. Constructed contingency tables are shown in table 2.5.

Table 2.5: Contingency Tables for Operators.

Operator A&B			**Operator A&C**		**Operator B&C**	
	A	**B**	**A**	**C**	**B**	**C**
Good sample	7955	8892	7955	8247	8892	8242
Total	8984	9653	8984	9430	9653	9430

Where:
A= AIRTEL GSM Operator
B=MTN GSM Operator
C=GLO GSM Operator
On the basis of these contingency tables, the hypothesis

H_{XY}, says that the performance of network X does not differ significantly and is not independent of network Y while H'_{XY} says that the performance of network X differ significantly and is independent of network Y. In the case of 2X2 contingency, the x^2 is calculated from the following expression.

$$X^2 = \frac{(Nx+Ny)\ (KyNx-KxNy)^2}{NxNy\ (Kx+Ky)\ (Nx+Ny-Kx-Ky)} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots..2.1$$

Where N denotes the total sample number and K denotes the good sample number for network operators X and Y. For instance, for the contingency table between network operator A and B, N_A represents the total sample size of network operator A (9880) while K_A represents the good sample size (7955). Thus, x^2 test are applied accordingly, and result provided in table 2.6.

TABLE 2.6 TEST RESULT FOR THE NETWORK

OPERATORS	X^2
AIRTEL & MTN	68.3
AIRTEL & GLO	0.0021
MTN & GLO	113.29

Now, for testing the hypothesis, a threshold for the test statistics is determined as 3.841 where chi-square with 1 degree of freedom at 0.95 is assumed (significance level of 0.05). Then, for example, for networks X and Y, H_{XY} is rejected if the test statistics exceeds this threshold. Accordingly to the test results in table 2.6 $H_{AIRTEL/\ MTN}$ and $H_{MTN/GLO}$ are rejected where as $H_{MTN/GLO}$ is accepted. Thus, it is concluded that the performance of MTN network operator is independent of the other two network operators. On the other hand, the performances of AIRTEL and GLO network operators are not significantly independent.

X^2 test is a reliable test in this research since the number of observations exceeds 5 in the above contingency tables, however, if the sample sizes are small, fisher's exact test could be used as a statistical significance test in the analysis of contingency tables.

2.7 FISHER'S EXACT TEST

Fishers test is one of the exact test as the significance of the deviation from a null hypothesis can be calculated exactly. Then the hypothesis tests where repeated using fishers exact test. Accordingly the probabilities P, for the contingency table are calculated from:

$$P = \frac{\begin{pmatrix} Nx \\ KX \end{pmatrix} \begin{pmatrix} Ny \\ K_Y \end{pmatrix}}{\begin{pmatrix} NX + NY \\ Kx + Ky \end{pmatrix}} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 2.2$$

And the results are listed in table 2.7

Table 2.7 Probabilities for the contingency table using Fishers test.

NETWORK OPERATORS	P
AIRTEL & MTN	0.52
AIRTEL & GLO	0.5
MTN & GLO	1

As can be seen from table 2.7, calculated probabilities are not less than the significance level (0.05). Then

$H_{AIRTEL\&\ MTN}$, $H_{AIRTEL\&GLO}$ and $H_{AIRTEL\&GLO}$ are not rejected.Thus, it is concluded that the performance of networks AIRTEL, MTN and GLO are independent. Then, according to x^2 test, all networks can be compared in pairs and if the performance differs significantly, they receive one point.

Summarily, the information gotten from the research conducted by the Nigerian communication commission may not been seen as being validated enough considering the fact the questionnaire method is not an acceptable scientific method of problem evaluation and analysis.

III. METHODOLOGY, DESIGN AND IMPLEMENTATION

3.1 RESEARCH METHODOLOGY

Quality of services (Qos) can be described as the ability of a network to provide a service at an assured service level. This can be measured by either the network operators by itself or by an independent or a regulatory organization.

In order to study the performance of the proposed method, different sets of data were generated using the Erlang B formula.
In this research work, finite numbers of subscribers were used in loading the network channel using the Erlang B formula. These numbers of subscribers were used together with various allocation time of service period of 60 seconds, 120 seconds, and 180 seconds respectively.
The range of subscribers chosen for this work includes 200, 250, and 280 numbers of subscribers.
The dimensioning was based on the loaded Erlang B formula.
3.2 Call Blocking : This is complete denial of a customer's call resource access to the network after a request has be made due to unavailable or busy channels.
The major causes of blocked calls is Network Congestion, and adjacent and co-channel interferences, under which every other factors or cause are attached to.
Each base station has certain fixed number of channel available to carry data traffic. If any new call arrives to a base station, it will first check for the availability of a free channel. If a free channel is are available, then this free channel is allocated for the call, but if there is no free channel available, then the call is blocked.
The amount of blocking can be reduced by changing or adjusting the capacity of the network in a way that more communication channels are made available in locations with higher user density.
In GSM Communication, resources are shared in terms of Time and Frequency i.e (Channelization Access Protocol).
Network dimensioning could be said to be the allocation of time and frequency to the users of a particular network channel to reduce congestion.
It is used to determine the amount of traffic the radio channel capture. An accurate traffic model for the system will greatly enhance the accuracy of network dimensioning, extending network investments and increasing the profit returns. When a network is properly dimensioned, the channels will be used more efficiently and will produce greater user satisfaction.
On the other side, poor modeling of network traffic characteristics can actually affect system performance. Improper dimensioning of network will lead to congestion in the cell, which will also lead to increased blocked calls, increased unsuccessful handovers and poor call accessibility and retainability.
Traffic planning focuses on the busy hour, which is for effective traffic management. For the purpose of this work, Network Dimensioning here relied on Erlang B formula for its mathematical models.
Network Accessibility:- This is the probability that a service can be obtained within a given conditions when requested by a service user. In other words, it measures the ease in which calls are established (i.e the call set up success rate (CSSR). It denotes that, the higher the value of CSSR, the easier it is to set up a all.
It is measured by the ratio of the number of successful calls (#SC) in the tests to the number of call attempts (#CA).

$$\text{i.e Accessibility} = \frac{\#SC}{\#CA} = (1\text{- Blocking probability}) \times 100\% \quad \text{.....3.1}$$

Call Setup Traffic Channel (TCH) Congestion rate: The call Setup Traffic Channel Congestion Rate provides the percentage of call attempts to allocate a TCH call setup that were blocked in cell.

$$\text{Call Setup TCH Congestion Rate} = \frac{\text{No of TCH blocking}}{\text{No of TCH attempts}} \quad \text{...... 3.2}$$

Possible reasons for call setup block could be:

- Increasing Traffic Demand.

- Bad Dimensioning.
- High Antenna Position.
- High Mean Holding Time.
- Low Handover Activity.
- Congestion in Surrounding Cell.

3.3*NETWORK CONJECTION*: Congestion is a phenomenon in telecommunication system that occurs when more subscribers attempts simultaneously to access the network than it is able to handle. This is a situation where subscriber's number has completely overgrown network capacity.

The problem of network congestion is a network managerial issue that affects the quality of service (Qos) rendered by a network, apart from the fact that over utilization of a node in network can lead to resources short span or malfunctioning. Therefore congestion control is of utmost importance for the sustainability of the system and the development of congestion control measure in this system should be a good effort in the right direction.

REASONS/CAUSES OF NETWORK CONGESTION ARE:

- **Lack of adequate infrastructure**: To guarantee efficient network quality, there must be adequate infrastructural equipments to be able to drive the network. Also, the size of these equipments must be in tandem with the subscriber's base. When subscriber's base overgrows infrastructural equipment, the congestion is inevitable. In Nigerian situation, operators have been playing down on expansion of all cell sites, which of course is the strength of call quality. The rate of service rollout in the country has never been the same with rate of infrastructural roll out, and this often leads to network congestion and inability to recharge phones.
- **Insufficient channels:** Since there are not enough infrastructural equipments (e.g base stations), automatically there will be lack of adequate network channels to support network functionality. Recall that channels are normally used to determine total number of subscribers that can be allowed to use a base station.
- **Improper network configuration:** The network configuration not being set up properly, such that one cell is not aware of the cell the phone is trying to handover to.

3.4 HANDOVER (HANDOFF): This is the process of transferring a call (or data transfer) in progress from one channel (base station) to another.

During handover processes, handoff calls could be blocked as a result of non availability of channel to accommodate such. This in turn could cause call dropping.

Handover success rate is one of the major key performance indicator (KPI) that should be optimized to improve handover quality. It shows the percentage of successful handovers of all handover attempts. A handover attempt is when a handover command is sent to the mobile .

Possible reasons for poor handover success rate could be:

- Congestion
- Poor Link Connection.
- Bad Antenna Installation .
- Incorrect Handover Relations.
- Incorrect Locating Parameter Setting.
- Bad Radio coverage.

REASONS/ CAUSES OF HANDOVER

- If the mobile device moves out of the range of one cell (base station) and a different base station can provide with a stronger signal.
- If all the channels of one base station are busy then a nearby base station can provide service to the device.

TYPES OF HANDOVERS

- Hard Handover (for GSM system)
- Soft handover (for CDMA systems)

N.B for a handover process to be successful means that there will not be a break of radio link for cell-to-cell transfer of call.

3.5 ERLANG B FORMULA

This formula could be used to dimension a GSM Network so as to increase the network retainability, call accessibility, handover success rate, thereby reducing the call block occurrences.

Some GSM network operators achieve the reduction of their drop calls at the expense of increased blocking probability. Traffic Engineering is key to effective wireless telephony, network design and planning. Traffic characteristics need to be addressed and once accomplished; mathematical models can be applied to dimension the network.

The network planning is built around three variables:

- **Servers**: This is the channel that handles calls.
- **Traffic:** This is the use of radio channels.
- **Grade of service:** The probability that all servers will be busy when a call attempt is made.

$$\text{Grade of service} = \frac{\text{Number of lost calls}}{\text{Number of calls}} \quad \ldots\ldots\ldots\ldots\ldots\ldots..3.3$$

Network dimensioning relied on Erlang B formula for its mathematical models.

ERLANG B FORMULA is used to calculate the probability that a resource request from the customer will be denied due to lack of resources. (i.e due to congested network or unavailable channels). Also, when the blocking probability is known, (which could be gotten using equation 3.1) then, means of increasing the number of channels could be achieved.

KEY CONCEPTS IN ERLANG B FORMULA.

Traffic: This is defined as the use of radio channels. When a user makes a phone call, a channel is seized for communication thereby generating traffic.

Busy Hour: The load handled by a system varies a lot based on the time of day and days of the week. Most systems are heavily loaded for a few hours in a day. This is the period when incoming calls are most likely to be blocked or turned away, so this is the time statistics are calculated.

The main objective of resource dimensioning is to make sure that the GSM network system performs well during those busy hours. This will make sure that the system has adequate resources to handle peak as well as off peak hour.

Blocking Probability: The blocking or outage probability is defined as the probability that a call is blocked. It is a measure of Grade of service. It can also be defined as the chance that a customer will be denied service due to lack of resources.

Blocking probability = 1-Network Accessibility………..3.4

For example, a blocking probability of 0.01 means that 1% of the customers will be denied service. Most of the time, blocking probability is calculated during the peak (busy) hour(s).

Grade of service: The percentages of incoming calls are turned away during the busy hours because all the lines are busy at the time of the calls. A higher grade of service guarantee to the customer means ensuring low blocking probability during the busy hours.

Number of Lines©: This denotes the number of independent phone lines or the numbers of customers that subscribe to the phone service. It determines the number of calls that are been serviced concurrently.

Service Time: The total time needed by a resource to handle one call.

Wait Time: The total time customers will have to want in the queue before they get any service.

Arrival rate (λ): This is the number of calls that arrive per unit time. If five hundred calls arrives in an hour, on the average, the arrival rate is 500 calls/hours.

Service Rate(μ): This is the mean number of calls serviced per unit time.

Total Traffic in Erlang ($L=\lambda/\mu$)

ERLANG is a unit of traffic measurement. It is the ratio of Arrival Rate to service rate: It is also the amount of time an average subscriber uses his phone during the so-called busy hour (i.e the hour of the day when we expect the highest number of calls).

3.6 ERLANG B FORMULA MATHEMATICAL MODEL

Earlang B formula is also said to be the probability that all channels (n) are busy. It gives the proportion of time that no new calls can enter the system.

ERLANG B formula model is stated below as:

$$P_{bl} = (\ell^n/n!) / \left(\sum_{i=0}^{n} \ell^i/i!\right) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots..3.5$$

where: $\ell =$ offered traffic in erlang

n= number of timeslot

P_{bl}= blocking probability

but, $\ell = (N \times M)/3600$……………………………………...3.6

where N= Number of lines or calls per hours

M= Duration of each call in seconds.

3.7 Loading Channel Using Erlang B Formula

For the purpose of this research work, three case studies are used using Erlang formula to determine the accepted loading for this model.

Loading: This refers to the total number of subscribers a channel is expected to serve: A loading of 200 subscribers means, the channel is expected to serve 200 subscribers.

For this work, we assumed a loading values of 200, 250, and 280. Also, we assumed a timeslot of 8, and a variable service time of 60 seconds, 120 seconds, and 180 seconds respectively. The loading technique was also repeated using the same values for the variables with a timeslot of 16, and MATLAB software as shown in chapter four and five were used to generate the simulation parameters in all the processes.

3.8 SIMULATING THE PROCESSES

Simulation One:

Inputting the following values into MATLAB using equations (3.5)and(3.6) respectively generated the parameters for simulation one as shown in table 4.1 and plotted the graph shown in figure 4.1(a&b).

i.e N = 200, 250, 280
n = 8
m = 60

Simulation Two:

Also by inputting the following values into MATLAB using equations (3.5)and(3.6) respectively generated the parameters for simulation two as shown in table 4.2 and plotted the graph shown in figure 4.2(a&b).

i.e N = 200, 250, 280
n = 8
m = 120

Simulation Three:

Also by inputting the following values into MATLAB using equations (3.5)and(3.6) respectively generated the parameters for simulation three as shown in table 4.3 and plotted the graph shown in figure 4.3(a&b).

i.e N = 200, 250, 280
n = 8
m = 180

Simulation four:

Inputting the following values into MATLAB using equations (3.5)and(3.6) respectively generated the parameters for simulation four as shown in table 4.4 and plotted the graph shown in figure 4.4(a&b).

i.e N = 200, 250, 280
n = 16
m = 60

Simulation five:

Inputting the following values into MATLAB using equations (3.5)and(3.6) respectively generated the parameters for simulation five as shown in table 4.5 and plotted the graph shown in figure 4.5(a&b).

i.e N = 200, 250, 280
n = 16
m = 120

Simulation six:

By inputting the following values into MATLAB using equations (3.5)and(3.6) respectively generated the parameters for simulation six as shown in table 4.6and plotted the graph shown in figure 4.6(a&b).

i.e N = 200, 250, 280
n = 16
m = 180

3.9 DEVELOPED MODELS FOR MANAGING NETWORK CONGESTION

Below shows the model that could be used to manage the congestion that might result from the loading model chosen or other unexpected sources.

a. **Block time sharing:** Here, each call is given a specific time of block and until the time expires, no other call can use the channel. Each call is given full access to the network system according to the time allocated but as soon as the time expired, a control is passed to pre-empt the call as to allow other callers to have access to the channel.

Advantages of this method are:

- It does not allow any call to a channel more than the allowed time.

- It also gives access to the call without interruption from any other call within the time limit.
- It allows equal sharing among the call.

Its DRAW BACK is that it wastes service time. For those calls that do not finish their allotted time before they exit the channel, the remaining time will be wasted.

Below shows the algorithm using pseudo code that could be used to achieve the process:

1. START
2. SET THE BLOCK TIME
3. DO WHILE SERVICE= TRUE
4. SET K=1 // INITIAIISING TIMESLOT
5. DO WHILE NOT FREE AND K<=8// TESTING FOR FREE TIMESLOT
6. R = K + 1
7. LOOP
8. IF TIMESLOT= FREE THEN
9. LOAD THE CALL TO TIMELSLOT (TK)
10. WHILE SERVICE TIME < BLOCK TIME OR SERVICE TIME < AIR TIME
11. CONTINUE THE CALL
12. ELSE
13. REJECT THE CALL
14. END IF
15. FETCH THE NEXT CALL
16. LOOP

(b) **Dynamic Allocation without time slicing**. This will allow calls to occupy the channel without given any time range. Any call that enters the channel will finish its work before allowing any other call to the channel. Also, it allows any call that is ready to seize the channel without any consideration.

The advantage of this method is that it allows the call to finish its work before any other call can be allowed to the channel.

The drawback is that, sometimes, some calls might occupy the channel unnecessarily thereby denying others from entering.

The following algorithm could be used to achieve this;

1. START
2. DO WHILE SERVICE= TRUE
3. SET K=1 // INITIAIISING TIMESLOT
4. DO WHILE NOT FREE AND K<=8// TESTING FOR FREE TIMESLOT
5. K = K + 1
6. LOOP
7. IF TIMESLOT= FREE THEN
8. LOAD THE CALL TO TIMELSLOT (TK)
9. WHILE SERVICE TIME <AIR TIME
10. CONTINUE THE CALL
11. ELSE
12. REJECT THE CALL
13. END IF
14. FETCH THE NEXT CALL
15. LOOP

(c**) Priority Allocation**

In this method, everybody should have a level of priority and this priority should have been integrated into the SIM card.

So, anytime anybody buys a SIM card and activate it on the network, the priority level registered automatically. The priority level will be used throughout the period of subscription of the subscriber to the network. The level of priority will be determined by the network of your service. This model will be able to take care for the executive essential duties officers like president, Governors, fire fighters, police and so on.

In this model, the principles of arithmetic operation preferences will be used where the highest priority will again access to the channel before the lower priority.

So this model is meant especially for essential duties officers. In this model, time interrupt pre-emption will not be applied; their access to the network will not be terminated until they terminate it themselves. The advantage of this model is that it will allow the essential duties officers calls to complete their calls without any interrupt and thereby forestalls causalities that might occur if they were not given attention. The DRAW BACK of this method is that many calls will be denied access during the time when the essential duties are on. Also, some calls that may be very important to some subscribers might be dropped and immediate attention may not be possible easily.

Below shows the algorithm to achieve that:

1.START
2. DO WHILE SERVICE= TRUE
3. SET ESSENTIAL CODES
4. SET K=1 // INITIAIISING TIMESLOT
5. DO WHILE NOT FREE AND K<=8// TESTING FOR FREE TIMESLOT
6. K = K + 1
7. LOOP
8. IF TIMESLOT= FREE THEN ESSENTIAL CALL
9. DROP CALL WITH LEAST PRIORITY
10. LOAD THE ESSENTIAL CALL TO TIMESLOT (TK)
11. WHILE SERVICE TIME > 0
12. CONTINUE THE CALL
13. ELSE
14. LOAD THE CALL TO TIMESLOT (TK)
15. WHILE SERVICE TIME > 0
16. CONTINUE THE CALL
17. WEND
18. END IF
19. FETCH THE NEXT CALL
20. LOOP

(d) Dynamic Allocation Time Slicing with Signal Sensing: This allows calls to occupy the channel with a maximum time interval. This will allow calls to occupy the channel with an attach time range. Unlike the block time-sharing, it will not preempt the calls automatically but a carrier sense will be passed at intervals to check if there is any call waiting. If there is any call waiting, then, the signal will preempt any call that had exhausted its maximum time allocation, other- wise, the call will be allowed to continue as long as the caller wishes to continue. The advantages are:

• It does not allow the system to be occupied unnecessarily.

• It allows the subscriber to continue calling as long as they wish provided there is no call waiting.

The **disadvantage** is that it does not consider any call as important thereby essential calls will be pre-empted without finishing the call. Below shows the Algorithm:

1. START
2. SET THE MAX_TIME
3. DO WHILE SERVICE = TRUE
4. SET K = 1// INITIALISING THE TIMESLOT
5. DO WHILE NOT FREE AND K<=8
6. //TESTING FOR FREE TIMESLOT
7. K = K+1
8. LOOP
9. IF TIMESLOT = FREE THEN
10. LOAD THE CALL TIMESLOT (TK)
11. WHILE SERVICE_TIME < MAX_TIME AND NO CALL WAITING OR SERVICE_TIME < AIR_ TIME CONTINUE THE CALL
12. WEND
13. ELSE
14. REJECT THE CALL
15. END IF
16. FETCH THE NEXT CALL
17. LOOP 5

(e)Hybrid Algorithm: This takes into account the strengths and constraints of all other network management Algorithms. The steps below are recommended as general order for introducing the different solutions in the network layer:

The first step is to determine the appropriate channel loading model for the network. Make sure the number of channels to subscribers does not result into the operator's deficit. Then the below algorithm should be used to manage calls within the network. The recommended algorithm has a prioritization scheduled to maintain un-interrupted communication during emergency. It also provides a location for temporary memory to cater for incessant frequent callers within a specified period. It gives them a higher priority over a new entrant call. This algorithm should be implemented at every base station. The advantages of this algorithm are:

• It gave a priority to highly essential duties calls that needed immediate attention. This will thereby forestall any casualties that may occur if such attention is not given.

• It gave priority to the most denied calls to grab the channel when they appear within a specified time.

• It does not allow any call to occupy the channel more than necessary when they are calls waiting to grab the channel.

• It does not preempt the subscriber if there is no call waiting unlike the block-time that will preempt even if there is no call waiting

• It allows dynamic allocation of channel when there is equal priority calls.

It allows dynamic allocation of channel when there is equal priority calls. RECOMMENDED ALGORITHMS:

1. START
2. DO WHILE SERVICE = TRUE
3. SET ESSENTIAL_CODES
4. SET K = 1// INITIALISING THE TIMESLOT
5. DO WHILE NOT FREE AND K<=8//TESTING FOR FREE TIMESLOT
6. K = K+1
7. LOOP
8. IF TIMESLOT <> FREE AND CALL = ESSENTIAL CALL THEN
9. DROP CALL WITH LEAST PRIORITY
10. LOAD THE ESSENTIAL CALL TO TIMESLOT (TK)
11. WHILE SERVICE_TIME> 0
12. CONTINUE THE CALL
13. WEND
14. ELSE
15. IF TIMESLOT = FREE AND CALL = ESSENTIAL CALL THEN
16. LOAD THE CALL TO TIMESLOT (TK)
17. WHILE SERVICE_TIME > 0
18. CONTINUE THE CALL
19. WEND
20. ELSE
21. IF TIMESLOT = FREE THEN
22. LOAD THE CALL WITH THE HIGHEST NUMBER OF PRESENCE
23. WHILE SERVICE_TIME< MAX_TIME AND NO WAIT SIGNAL OR SERVICE_TIME<AIR_TIME
24. CONTINUE THE CALL
25. WEND
26. ELSE
27. REJECT THE CALL
28. REGISTER AT THE TEMPORARY MEMORY
29. END IF
30. END IF
31. END IF
32. FETCH THE NEXT CALL
33. LOOP

The above network management models if used will enhance the network retainability, Network accessibility and will bring a reduction in network congestion and block call rate.

IV. DATA PRESENTATION AND ANALYSIS

4.1 DATA AND VARIABLE PRESENTATION

In this work, variables such as allotted time of calls, number of chosen finite subscribers, probabilities of call blocking, and offered traffic in erlang were used in the simulation processes in conjunction with their respective values as generated from Erlang B formula.

Here, high number of subscribers were chosen based on the fact that network congestion mainly occurs when there is a huge number of subscribers or customers trying to have access to the network simultaneously, probably in the availability of limited channels.

4.2TABLES SHOWING SIMULATION PARAMETERS

Table 4.1: Showing parameters for simulation one

Allotted time for call(s)	No of finite subscribers chosen (N)	Probability of call blocking (P_{bl})	Timeslot chosen (n)	Offered traffic in erlang (ℓ)
60 (sec)	200	0.0136	8	3.3333
60 (sec)	250	0.0359	8	4.1667
60 (sec)	280	0.0551	8	4.6667

Table 4.2: Showing parameters for simulation two

Allotted time for call(s)	No of finite subscribers chosen (N)	Probability of call blocking (P_{bl})	Timeslot chosen (n)	Offered traffic in erlang (ℓ)
120 (sec)	200	0.1597	8	6.6666
120 (sec)	250	0.2539	8	8.3333
120 (sec)	280	0.3061	8	9.3333

Table 4.3: Showing parameters for simulation three

Allotted time for call(s)	No of finite subscribers chosen (N)	Probability of call blocking (P_{bl})	Timeslot chosen (n)	Offered traffic in erlang (ℓ)
180 (sec)	200	0.3383	8	10.0000
180 (sec)	250	0.4410	8	12.5000
180 (sec)	280	0.4905	8	14.000

Table 4.4: Showing parameters for simulation four

Allotted time for call(s)	No of finite subscribers chosen (N)	Probability of call blocking (P_{bl})	Timeslot chosen (n)	Offered traffic in erlang (ℓ)
60 (sec)	200	3.9902e-007	16	3.3333
60 (sec)	250	6.2837e-006	16	4.1667
60 (sec)	280	2.3898e-005	16	4.6667

Table 4.5: Showing parameters for simulation five

Allotted time for call(s)	No of finite subscribers chosen (N)	Probability of call blocking (P_{bl})	Timeslot chosen (n)	Offered traffic in erlang (ℓ)
120 (sec)	200	0.0012	16	6.6666
120 (sec)	250	0.0114	16	8.3333
120 (sec)	280	0.0340	16	9.3333

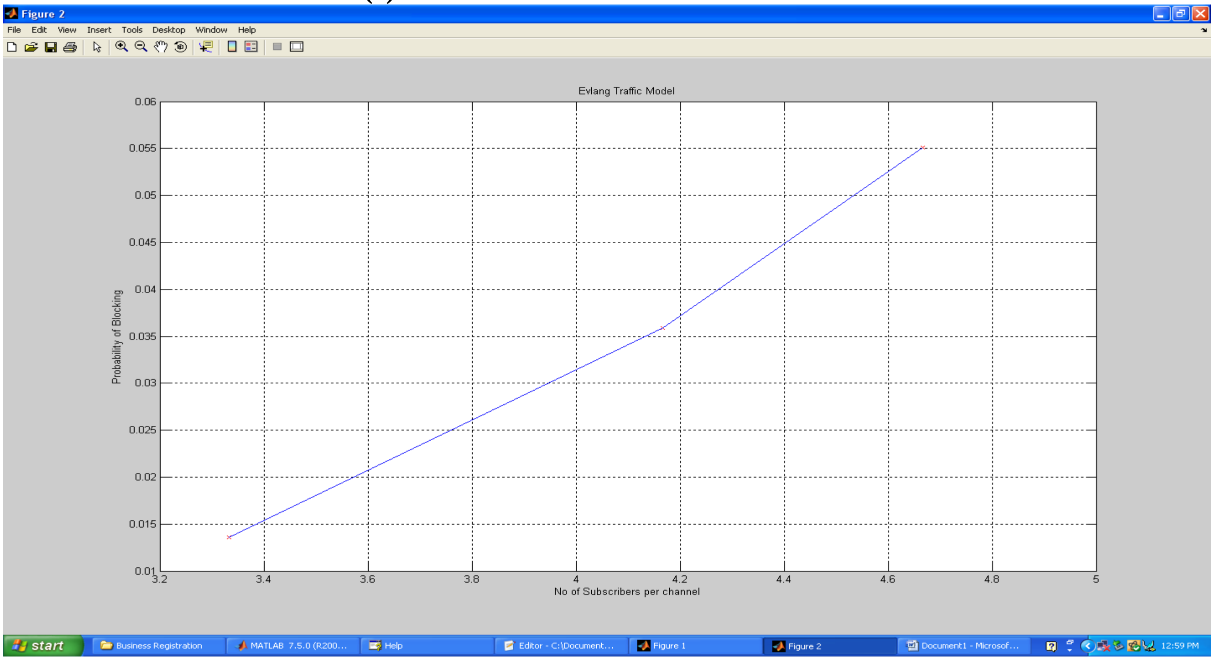
Table 4.6: Showing parameters for simulation six

Allotted time for call(s)	No of finite subscribers chosen (N)	Probability of call blocking (P_{bl})	Timeslot chosen (n)	Offered traffic in erlang (ℓ)
180 (sec)	200	0.0652	16	10.0000
180 (sec)	250	0.5066	16	12.5000
180 (sec)	280	1.3949	16	14.000

4.0 FIGURES SHOWING THE SIMULATION RESULTS.

Figure 4.1 a&b: showing the result for simulation one

(a)



(b)

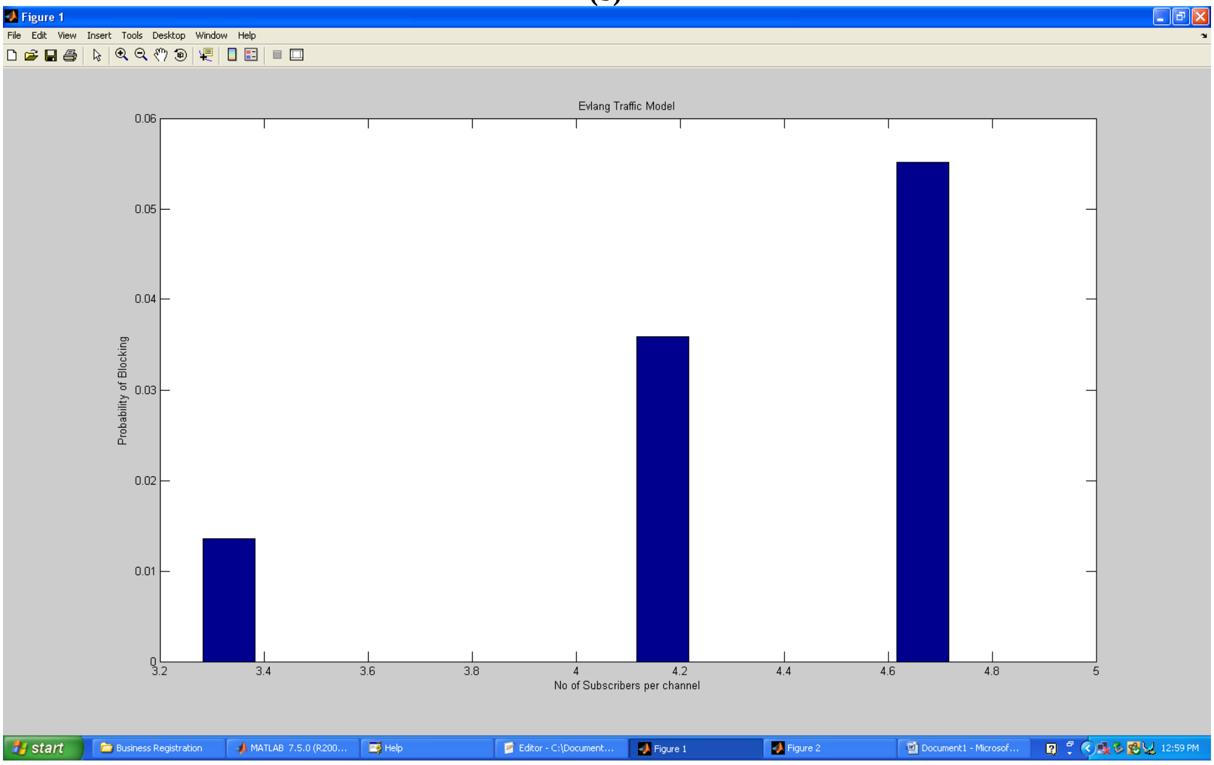
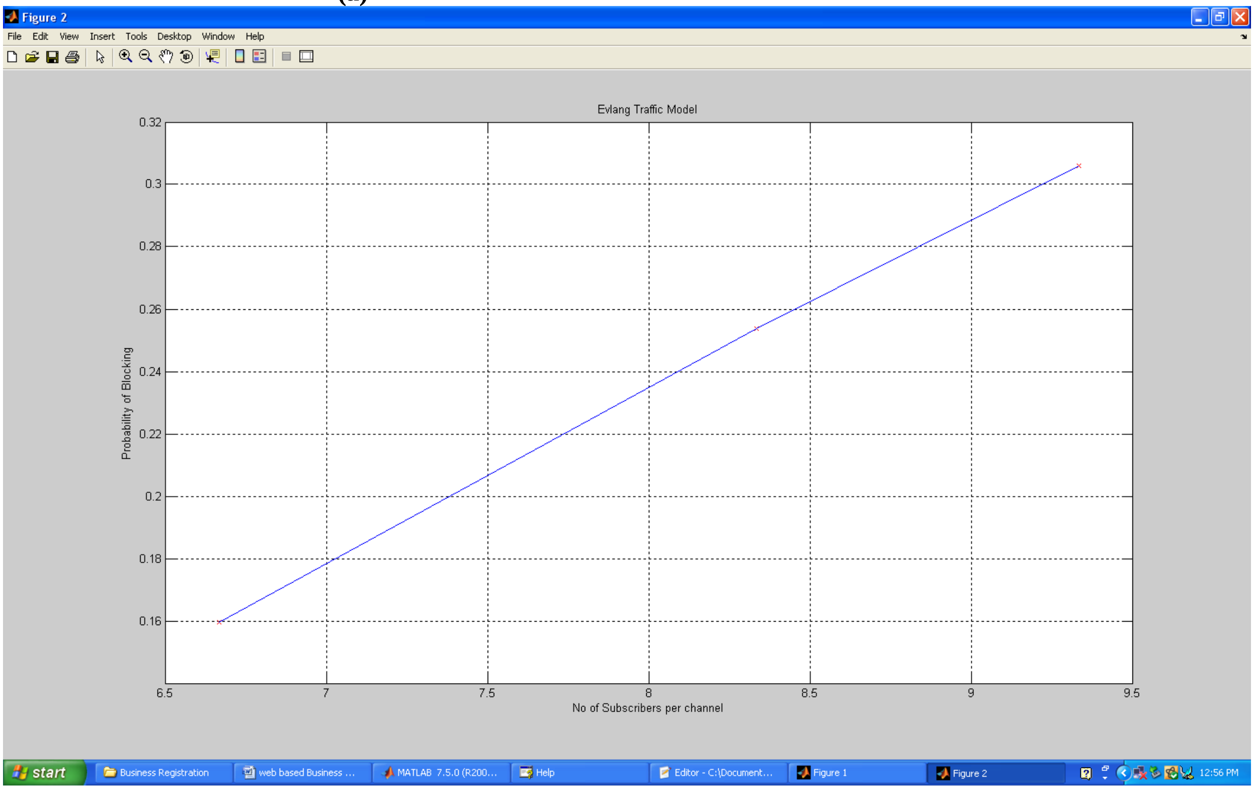


Figure 4.2 a&b: showing the result for simulation two

(a)



(b)

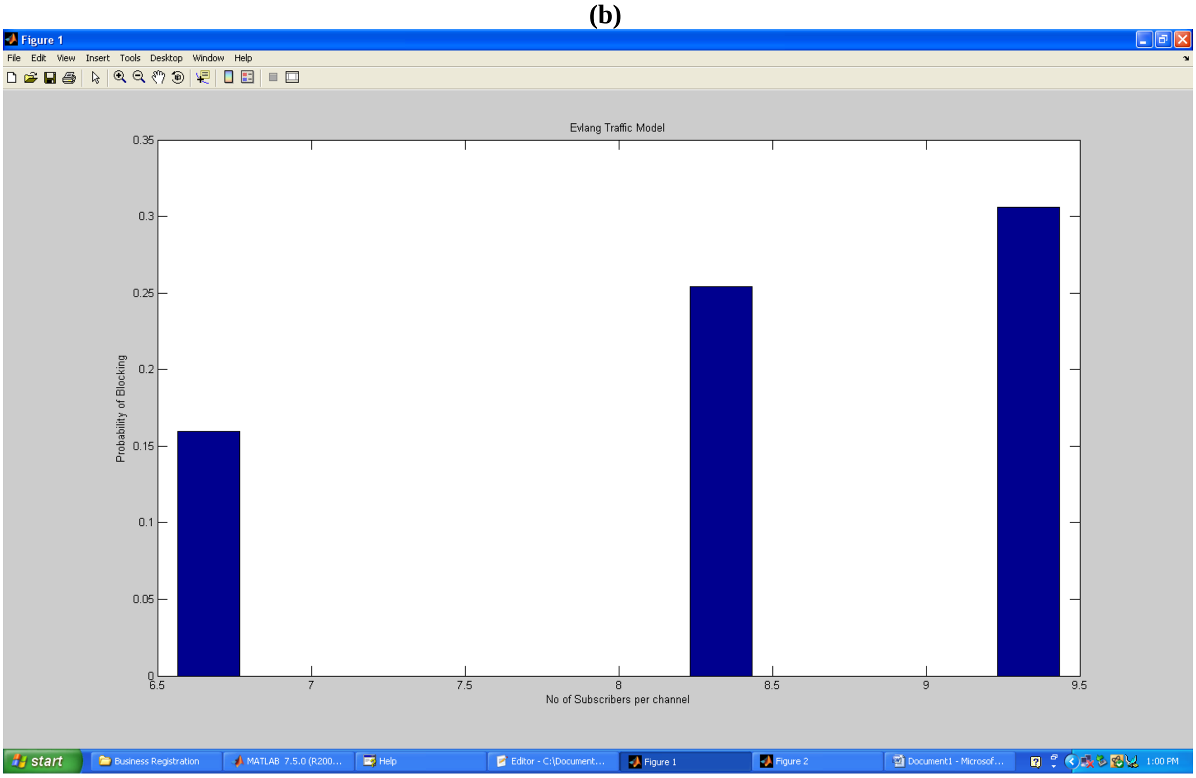
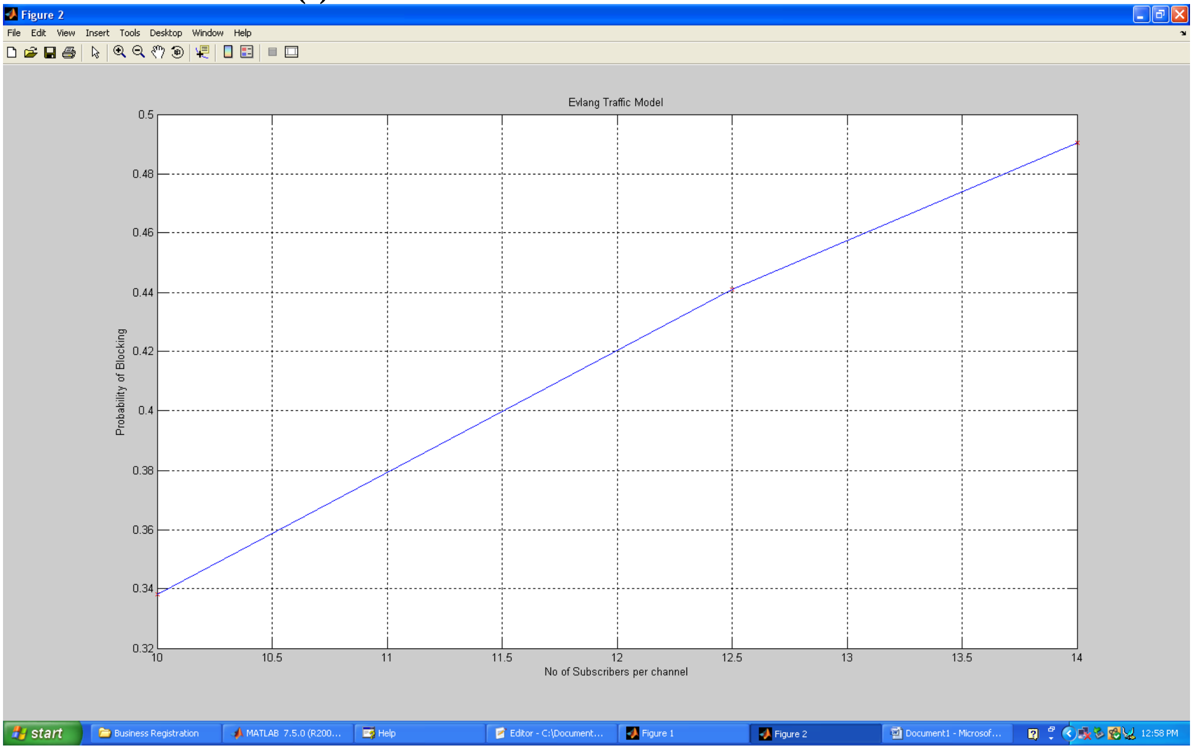


Figure 4.3 a&b: showing the result for simulation three

(a)



(b)

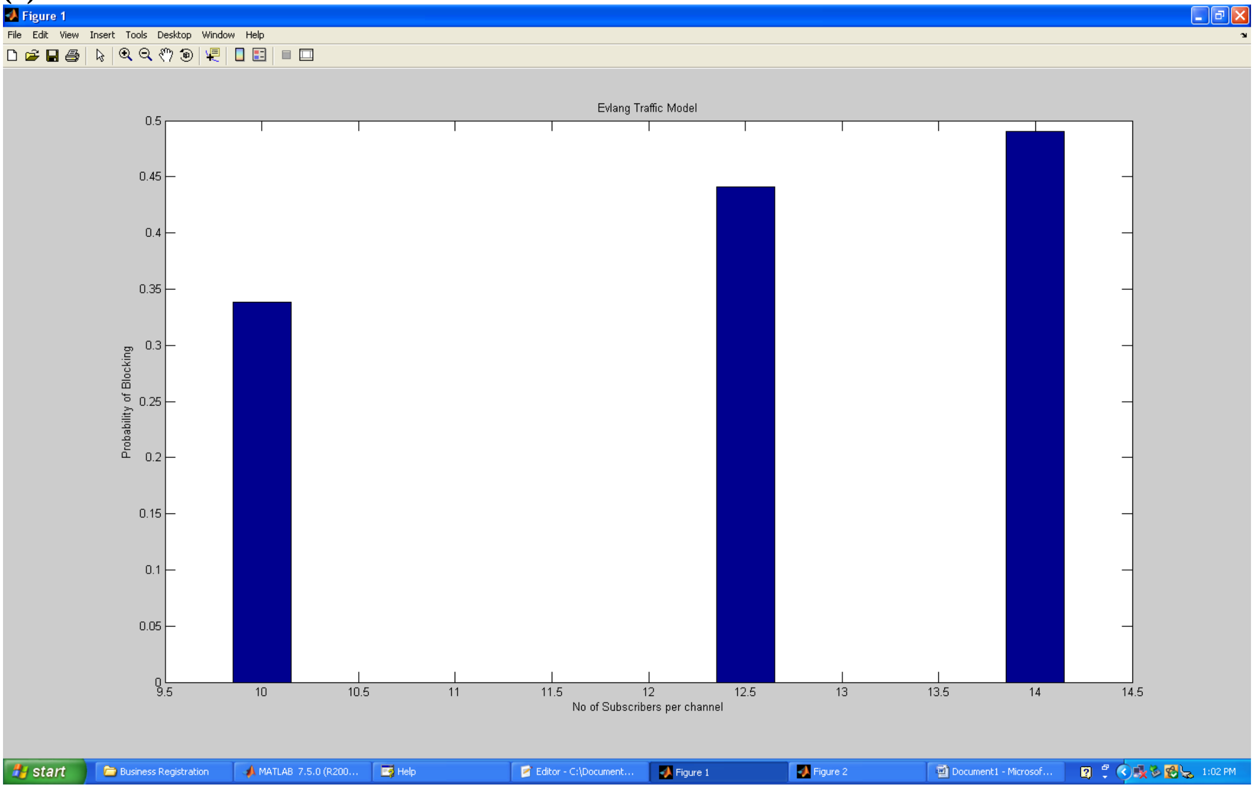
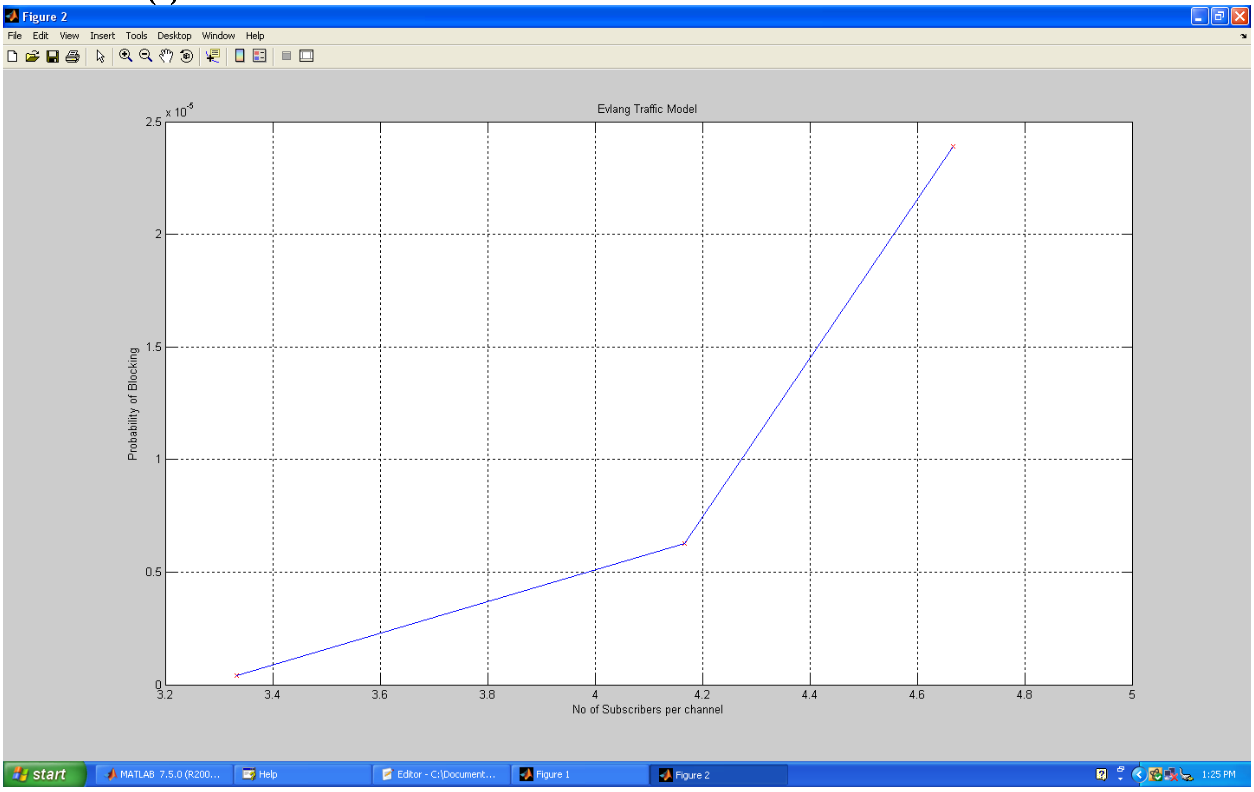


Figure 4.4 a&b: showing the result for simulation four

(a)



(b)

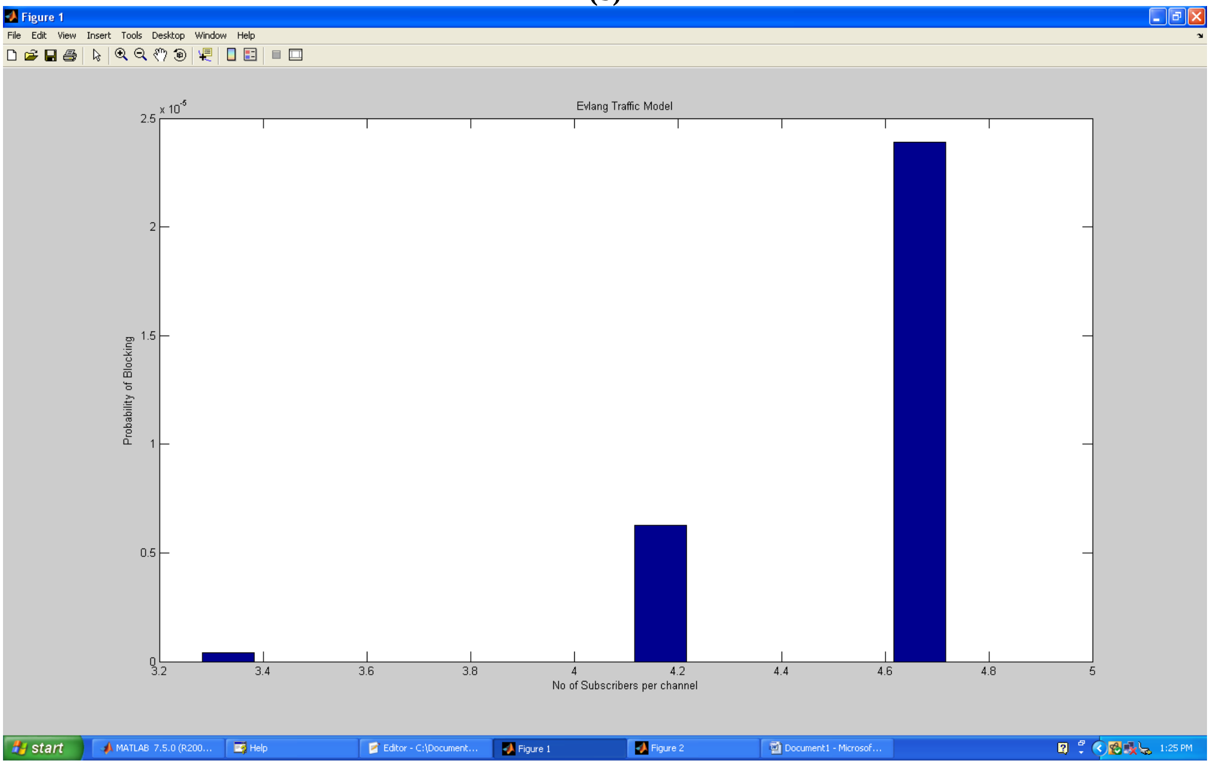
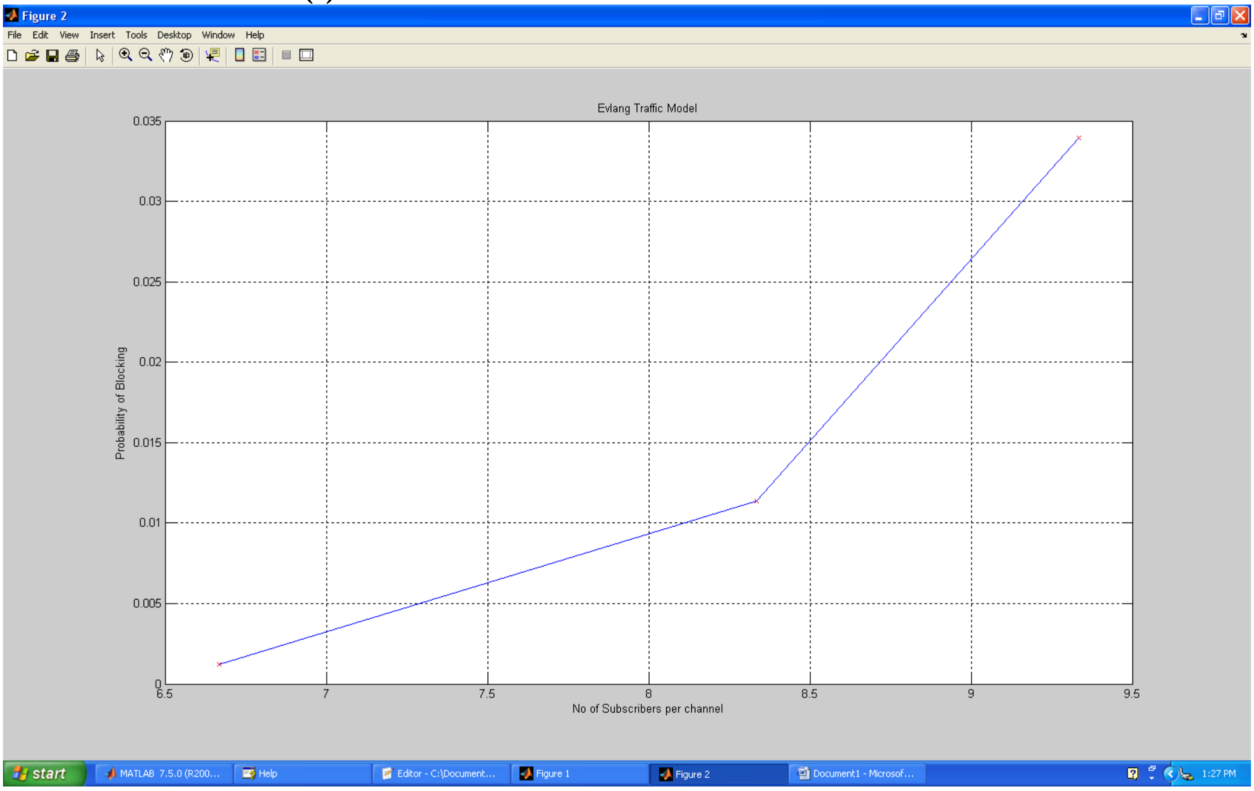


Figure 4.5 a&b: showing the result for simulation five

(a)



(b)

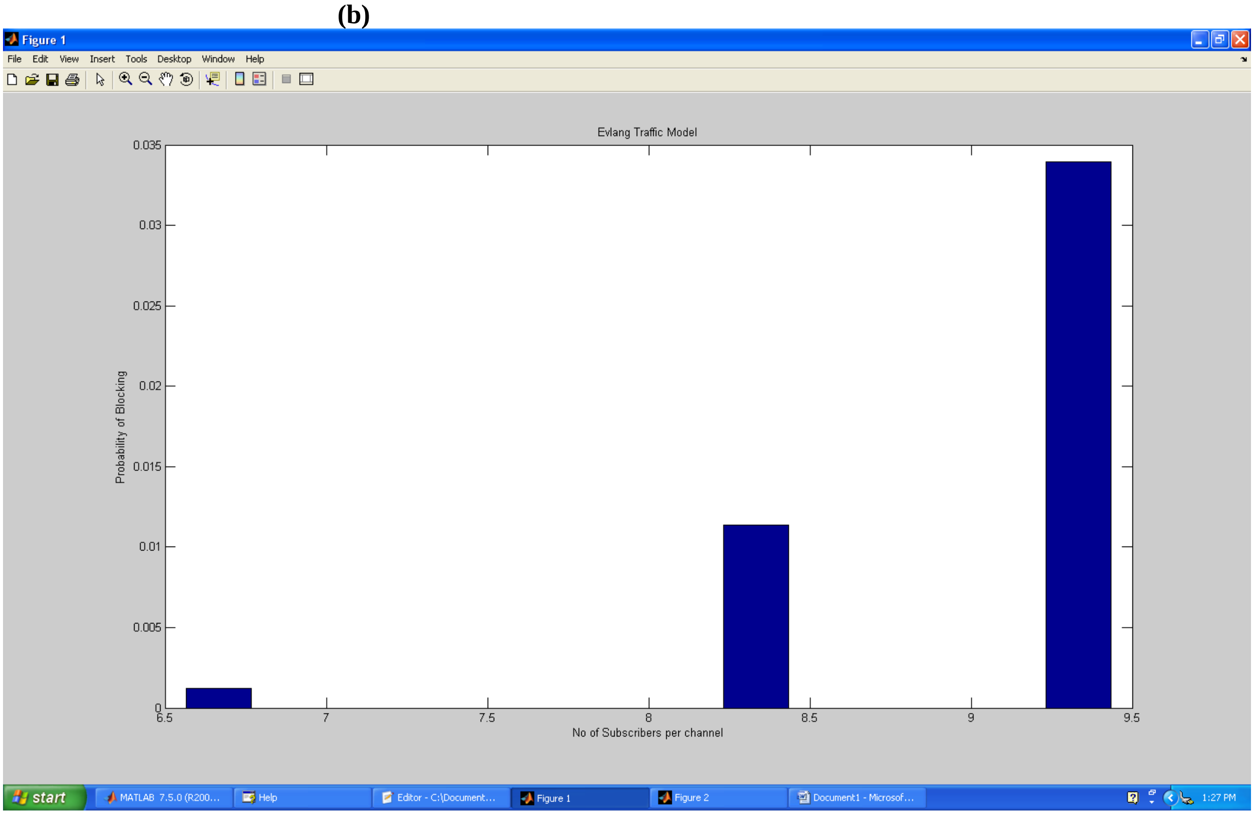
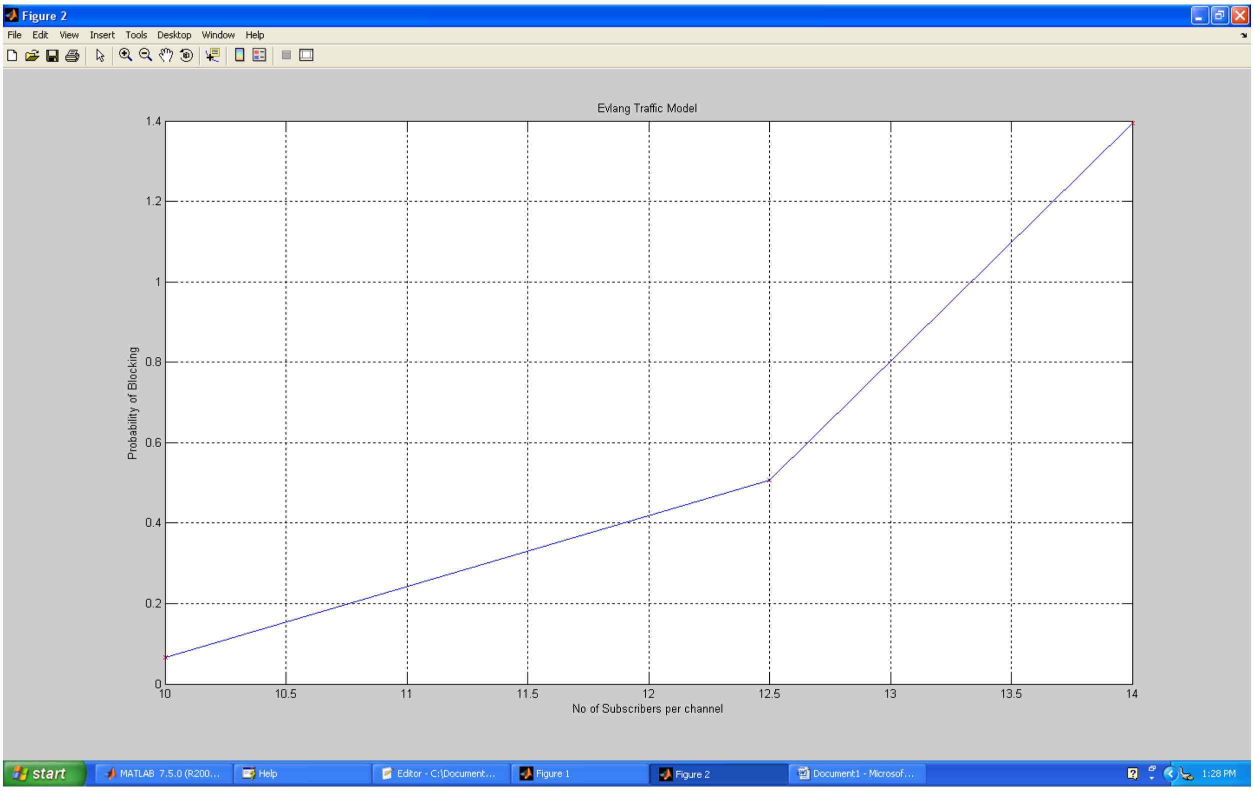
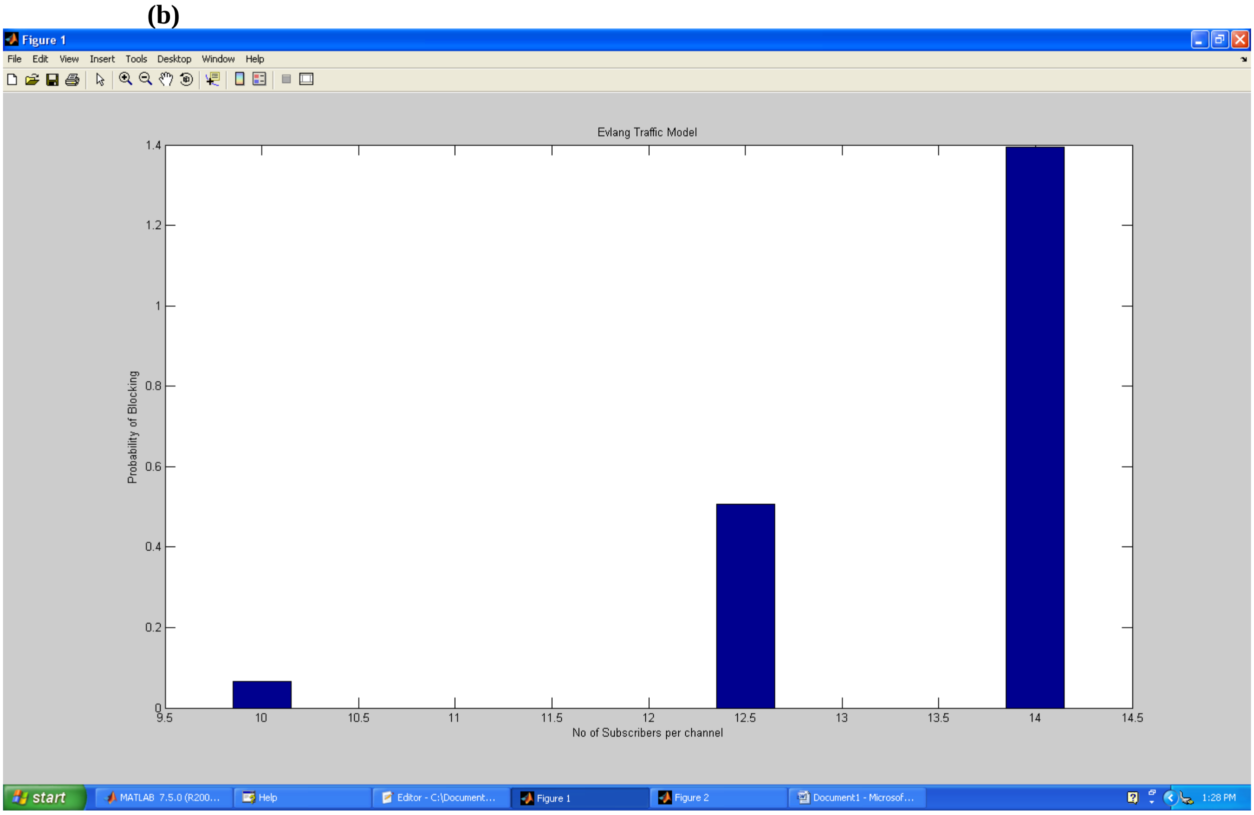


Figure 4.6 a&b: showing the result for simulation six

(a)



(b)



4.4 DISCUSSION AND FINDINGS BASED ON THE SIMULATIONS

The result of the simulation one indicates that 1% of the 200 subscribers will be denied access, 4% of 250 subscribers will be denied access and 6% of 280 subscribers will be denied access under the same condition of 60seconds maximum time.

The Erlang B performs excellently with the number of calls during the busiest hour, but here, we are dealing with following:

- Finite numbers of subscribers that are potential callers and net highest number of subscribers that are potential callers and not the highest number of subscribers that call during a particular hour. Every call is given equal time sharing.
- At any point in time, not more than 10% of the subscribers will be ready to call at any point in time.
- Out of those that are ready to call, not more than 40% will be ready to spend more than 1 minute considering the traffic of GSM services in Nigeria.
- We can see that congestion and call block be minimized with this type of model.

The operators should consider the cost of equipment, installation, maintenance, profit returns and the location of the base station when choosing any of the above loading case studies.

Taking all factors into consideration, I suggest the case study or simulation study one for the operators. The reason for the choice is that for example, for every 250 calls, only ten calls will be denied as against the other situation.

So with the first (simulation ones) case, there will be minimal congestion and a good profit returns.

Also, it is clearly from the simulation results of four to six, the chances of occurrence of call block is drastically reduced as a result of increased timeslot which depicted the availability of more channels, and also recorded a notable high probability of occurrence at every peak value of callers or subscribers using the same number of timeslot.

It is almost impossible to prevent network congestion due to some unexpected events, such as accidents and adverse.

Also from the simulation result of two and three, it is obvious that the probability of blocked calls increase drastically with respect to the offered traffic in erlang as a result of increase in the time allotted to the subscribers which shows that a minimal time show be allocated to subscribers to minimized the chances of blocked call occurrences unless the congestion control algorithm is being used to accommodate as many subscribers as possible using the priority time sharing, block time sharing, dynamic allocation without time slicing and or hybrid congestion technique.

If simulations two and three are be chosen, the gain of the operator might be reduced as a result of lost calls and more number of subscribers will be denied access.

It is almost impossible to prevent network congestion due to some unexpected events, such as accident and adverse weather conditions.

V. CONCLUSION, CONTRIBUTION AND RECOMMENDATION

5.1 CONCLUSION

Network performance is the most important parameter for measurement of quality of service. Poor performance of a telecom network would induce customer complaints and faults, thereby leading to customer dissatisfaction towards the operator. It is evident from this presentation that the quality of service rendered by these operators is far below expectation. So, urgent and proactive actors should be taken by the operators towards improving network performance. If this is done, customer could enjoy the best quality of service in terms of call success rate, voice quality, etc.

Customer satisfaction is critical to gain sustainable competitive edge in the GSM Market. In communication networks, as the customer satisfaction on the services is directing dependent on the quality and the performance of the network performance and Quality of Service (Qos) assessment are crucial. Also, from the research conducted by NCC and others in quest for ameliorating the call block problem, it was noted that from the average number of subscribers that dialed up to three or more times before getting connected, 58% of call blocking rate was recorded which after using more number of timeslots (which depicted more available channels) in this work was reduced to 28%, as showed in the simulation results and will also be reduced the more using the developed congestion control algorithm technique.

For this purpose, network operators should survey performance of their networks and measure quality parameters on a regular basis as customers meets and satisfaction are presumably to be the main market drive especially wide are service network such as cellular communications network.

As network optimatization engineers spend efforts to increase quality and capacity of operational networks, and to develop and deploy new services in order to meet custom demands and also guarantee customer satisfaction key performance indicators (KPIs) are to be kept within some specified threshold values in order to provide the Qos criteria required by both competent authorities and customers, since they are universally accepted parameters of cellular networks.

5.2 CONTRIBUTION

This research and study provides the following contributions:

1. Customer satisfaction and/or needs are integrated into benchmarking process and so, the quality of service assessment in order to see how customers perceive quality of service.
2. Acknowledgement of the fact that some unexpected events such as accidents and adverse weather condition can cause network congestion.
3. Introduces the use of algorithm in controlling network congestion

5.3 RECOMMENDATION

With these findings, it can be concluded that the Qos and overall performance of the GSM operation in Nigeria is poor, unreliable and unsatisfied. It is an indication that Nigerians are yet to really enjoy the impact of GSM as a new effective means of telecommunication.

In order to correct this ugly situation in the country and other countries with similar situation, suggestions on how to improve the Qos of the GSM operation in the country need to be made. It is on this basis that the following recommendations are made in order to ameliorate the observed defects:

1. The GSM operators in the country should focus more in building more BTS in order to increase their network coverage rather than the current competitions or bonanzas they are doing in order to win more customers.
2. NCC will be advised to inspect the GSM networks in the country regularly. This will aid the GSM operators in improving their networks base to meet their ever increasing subscriber's base. By this, the network accessibility in the country will improve while high congested networks currently experience shall be reduced.
3. Network operators should build additional switching centers across the country and increase capacity to handle traffic.
4. If a particular base station is to be taken "off line" (either for schedule maintenance, repairs, upgrades etc),all neighboring base station should have their communication level increased. This will increase their coverage area, thereby reducing congestion and dropped calls.
5. Operators should invest heavily in transmission network development and have a proper radio planning. This would ensure increased network resilience, improved bandwidth utilization and alleviation of capacity bottleneck.

It is believed that if the above recommendations are adhered to, the Qos and the overall performance of the GSM operation in the country shall definitely improve. Also, it will indeed improve the communication systems of the country as well as increasing revenue of the government, the citizenries and the operators.

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APPENDIX

```
clc
fprintf('Improving quality of service in GSM by reducing probability of call blocking through network\n\n')
fprintf(' dimensioning using Erlang b model and congestion control algorithm\n')
fprintf('      Presented by: Ahuchaogu Nnamdi  \n\n\n');
rmx =460;

ct =input ('Press Enter key to Continue ');

m=60
n=8
nlines=200
l= (nlines * m)/3600
l1=l
sum = 0;
for i=0:8

 sum = sum +( l^i/factorial(i))

end

pbl = ((l^n)/factorial(n)) / sum

nlines=250
l= (nlines * m)/3600
l2=l
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sum = 0;
for i=0:8

 sum = sum +( l^i/factorial(i))

end

pbl2 = ((l^n)/factorial(n)) / sum

nlines=280
l= (nlines * m)/3600
l3=l
sum = 0;
for i=0:8

 sum = sum +( l^i/factorial(i))

end

pbl3 = ((l^n)/factorial(n)) / sum

bx =[l1 l2 l3];
by=[pbl, pbl2, pbl3];
figure(1)
bar(bx, by, 0.2)
title 'Evlang Traffic Model'
ylabel('Probability of Blocking')
xlabel('No of Subscribers per channel')
figure(2)
plot(bx,by,'xr')

hold on
grid on
title 'Evlang Traffic Model'
ylabel('Probability of Blocking')
xlabel('No of Subscribers per channel')
plot(bx,by,'-b')
```

FOR SIMULATION ONE:
m = 60
n =8
nlines =200
l =3.3333
l_1 =3.3333
Sum =1
Sum = 4.3333
Sum = 9.8889
Sum =16.0617
Sum =21.2058
Sum =24.6351
Sum =26.5403
Sum =27.4476
Sum = 27.8256
pbl = 0.0136

nlines =250
l =4.1667
l2 =4.1667
Sum =1

Sum =5.1667
Sum =13.8472
Sum = 25.9035
Sum =38.4622
Sum =48.9278
Sum = 56.1955

sum = 60.5216
sum = 62.7747
pbl2 =0.0359

nlines =280
l = 4.6667
l3 = 4.6667
Sum = 1
Sum =5.6667
Sum = 16.5556
Sum =33.4938
Sum =53.2551
Sum = 71.6990
Sum =86.0443
Sum =95.6078
Sum =101.1865
pbl3 = 0.0551

FOR SIMULATION TWO:
m =120
n = 8
nlines = 200
l =6.6667
l1 =6.666
sum =1
sum = 7.6667
sum = 29.8889
sum = 79.2716
sum = 161.5761
sum =271.3155
sum =393.2481
sum =509.3744
sum = 606.1464
pbl =0.1597

nlines = 250
l = 8.3333
l2 =8.3333
sum =1
sum =9.3333
sum = 44.0556
sum = 140.5062
sum =341.4450
sum =676.3429
sum =1.1415e+003
sum = 1.6952e+003
sum =2.2720e+003
pbl2 =0.2539

nlines =280
l =9.3333

l3 =9.3333
sum =1
sum =10.3333
sum =53.8889
sum = 189.3951
sum =505.5761
sum =1.0958e+003
sum =2.0139e+003
sum =3.2380e+003
sum = 4.6662e+003
pbl3 =0.3061

FOR SIMULATION THREE:
m = 180
n = 8
nlines =200
l =10
l1 =10
sum =1
sum =11
sum =61
sum = 227.6667
sum =644.3333
sum =1.4777e+003
sum =2.8666e+003
sum =4.8507e+003
sum =7.3308e+003
pbl =0.3383

nlines =250
l =12.5000
l2 =12.5000
sum =1
sum =13.5000
sum =91.6250
sum =417.1458
sum = 1.4344e+003
sum =3.9775e+003
sum =9.2757e+003
sum = 1.8737e+004
sum = 3.3520e+004
pbl2 =0.4410

nlines =280
l = 14
l3 = 14
sum =1
sum =15
sum =113
sum =570.3333
sum =2171
sum =6.6529e+003
sum =1.7111e+004
sum = 3.8026e+004
sum = 7.4628e+004
pbl3 =0.4905

FOR SIMULATION FOUR
m = 60

n =16
nlines =200
l =3.3333
l1 = 3.3333
sum = 1
sum = 4.3333
sum = 9.8889
sum = 16.0617
sum = 21.2058
sum =24.6351
sum = 26.5403
sum = 27.4476
sum = 27.8256
pbl = 3.9902e-007

nlines = 250
l = 4.1667
l2 = 4.1667
sum =1
sum = 5.1667
sum = 13.8472
sum = 25.9035
sum = 38.4622
sum = 48.9278
sum = 56.1955
sum = 60.5216
sum = 62.7747
pbl2 = 6.2837e-006

nlines = 280
l =4.6667
l3 = 4.6667
sum = 1
sum = 5.6667
sum = 16.5556
sum = 33.4938
sum = 53.2551
sum = 71.6990
sum = 86.0443
sum = 95.6078
sum = 101.1865
pbl3 = 2.3898e-005

FOR SIMULATION FIVE

m = 120
n =16
nlines = 200
l = 6.6667
l1 = 6.6667
sum =1
sum = 7.6667
sum = 29.8889
sum = 79.2716
sum = 161.5761
sum = 271.3155
sum = 393.248
sum = 509.3744
sum = 606.1464

pbl = 0.0012

nlines = 250
l = 8.3333
l2 = 8.3333
sum = 1
sum = 9.3333
sum = 44.0556
sum = 140.5062
sum = 341.4450
sum = 676.3429
sum = 1.1415e+003
sum = 1.6952e+003
sum = 2.2720e+003
pbl2 = 0.0114

nlines = 280
l = 9.3333
l3 = 9.3333
sum = 1
sum = 10.3333
sum = 53.8889
sum = 189.3951
sum = 505.5761
sum = 1.0958e+003
sum = 2.0139e+003
sum = 3.2380e+003
sum = 4.6662e+003
pbl3 = 0.0340

FOR SIMULATION SIX

m = 180
n = 16
nlines = 200
l = 10
l1 = 10
sum = 1
sum = 11
sum = 61
sum = 227.6667
sum = 644.3333
sum =1.4777e+003
sum = 2.8666e+003
sum = 4.8507e+003
sum =7.3308e+003
pbl = 0.0652

nlines = 250
l = 12.5000
l2 = 12.5000
sum = 1
sum = 13.5000
sum = 91.6250
sum = 417.1458
sum = 1.4344e+003
sum = 3.9775e+003
sum = 9.2757e+003

sum = 1.8737e+004
sum = 3.3520e+004
pbl2 = 0.5066

nlines = 280
l = 14
l3 = 14
sum = 1
sum = 15
sum = 113
sum = 570.3333
sum = 2171
sum = 6.6529e+003
sum = 1.7111e+004
sum = 3.8026e+004
sum =7.4628e+004
pbl3 = 1.3949