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Review Article

Congestion and service quality improvement of mobile telephone networks in Nigeria: A review

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Abstract: Telecommunications and the ability to connect and interact has become a fundamental part of how the society operates. It is now the foundation for families, communities, businesses, and governments to connect and seamlessly share information in today's digital ecosystem. Mobile telephone network enables wireless communication using devices which include mobile phones, smart tablets. The mobile telephone networks provide the necessary infrastructure which are operated by service providers. The networks are usually not without congestion which is caused by a number of factors, including traffic build-up during peak periods and at hot spots, insufficient network dimensioning, especially in the switching system, and network operators' advertising offers. The consequence of network congestion is that an increase in offered load leads to either a small increase in network throughput or even a decrease in the throughput. In this work, congestion of mobile telephone networks has thus been reviewed. Improvement of Quality of Service (QoS) has also been examined. The congestion as a results of irregularity of public power supply, terminal breakdown, etc. are given sufficient considerations, while inadequate dimensioning of the network, particularly the switching and base station system and transmission impairment are not being adequately considered.

Keywords: Congestion, Quality of Service (QoS), Network, Mobile Telephone, Improvement.

1. INTRODUCTION

A mobile network, also referred to as cellular network, is a communication network or radio frequency network that is spread over given areas known as cells, each of which is served by at least one fixed-location transceiver. A transceiver, usually known as base station (BS) or a cell site, communicates wirelessly using the principle of radio signals, involving electromagnetic radiation. These signals are considered to be transverse waves in that they have a frequency and a wavelength. When these cells are linked together, they provide radio coverage over a large geographic region (Khan & Mauri, 2013). The base stations provide the cell with the network coverage which can be used for voice, data, and other types of content transmissions. To avoid interference and provide guaranteed service quality within each cell, a cell typically uses a different set of frequencies from neighboring cells.

The common types of cellular network include Global System for Mobile Communications (GSM) cellular network, Code Division Multiple Access (CDMA) cellular network and General Packet Radio Service (GPRS) cellular network. Others include Enhanced Data Rates for GSM Evolution (EDGE) cellular network and MOBITEX cellular network. Attention is paid to GSM and CDMA cellular networks in this work.

For GSM, cellular phones connect by searching for cells in the immediate vicinity. GSM phones may be identified by the presence of a Subscriber Identification Module (SIM) which contains a user's subscription information. This enables a user to switch devices while experiencing no reductions in service. GSM supports voice calls and data transfer. Advanced data services are permitted in GPRS. Data transmission is significantly faster than GSM (up to

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54kbps). It adds packet-switched functionality and the Internet on mobile handsets. The higher data rates allow users to take part in video conferences and interact with multimedia Web sites.

The CDMA is a form of multiplexing allowing many signals to occupy a single transmission channel, optimizing the use of available bandwidth. CDMA provides better capacity for voice and data communications. It provides data rate of up to 300 kbps and is the common platform on which 3G technologies are built. However, it is more expensive compared to GSM and GPRS. EDGE is an extension to the GSM/GPRS networks supporting higher data rates with average throughput of 70 to 130 kbps on both the downlink and the uplink.

Nonetheless, these networks usually experience congestion leading to poor quality of service (QoS). The rapid rise in the number of mobile subscribers has had an effect on the cellular network system's performance and the QoS (Popoola, Atayero, Faruk & Badejo, 2018). Network congestion is the reduced quality of service, and occurs when the network node or link is carrying or exchanging more data than it can comfortably handle. It is caused by a number of factors, such as bandwidth, traffic, antiquated devices, etc., resulting in overloading of the channel with packets that exceed its capacity. Bandwidth is, perhaps, the most common causes of network congestion. It is the maximum rate that data can pass through a given path. In other words, when there is no enough bandwidth to handle the amount of traffic for a given network, there is network congestion.

Congestion could also occur as a result of public power outages, terminal breakdowns, and other factors. Congestion occurs both in voice network and data network. For voice network, congestion occurs when a channel's potential is exceeded by the amount of calling operation or message traffic. Congestion occurs in data networks when the number of packets being transmitted through the network exceeds the capacity that the network's packet can handle. Some typical effects of congestion include packet loss, queueing delay and blocking of new connections.

Studies on congestion has been conducted by many researchers including (Shi et al., 2019; "'Boddu *et al., 2020)*. Congestion at peak hours and in hot spots may be a major issue. The peak period can be a time during the day when the number of packets transmitted across the network is at its highest. Hot spots are areas where high-speed wireless internet connectivity is accessible. Hotels, libraries, airports, cafes, and other public places are examples of such places. It appears that an increase in data rate is a natural solution to the present situation. However, due to the high cost of infrastructure that would be required, it will not be economically viable. Since traffic demand is rising regularly, the congestion issue will linger. This is why it is necessary to address it effectively (Chinecherem *et. al*., 2015).

As mentioned, there are different causes of congestion. But the most common causes of congestion in mobile telecommunications networks include inadequate network dimensioning, especially in the switching and base station subsystem (BSS) networks, which is a section of a traditional cellular telephone network responsible for handling traffic and signaling between the network switching subsystem and a mobile phone; a public power supply that is epileptic; fiber cut in the case of a fiber optic connection; intrusion and uneven terrain which are examples of propagation impairments; network operators promoting sales, such as free midnight calls, lower tariff promo, etc., and breakdown of terminals.

Since the minimally available network resources are shared by many end users, it is impossible for all end users to be connected to the network at once. Subscribers who are unable to set up a call are therefore queued for the next available or completed call channel (Uttam, 2014). As a result, it is often assumed that the network operators have failed in their duty to upgrade channels and integrate them into the core network in order to satisfy the growing customer base. The trend in mobile telephone services demand continuous channel expansion in order to offer outstanding coverage to existing and incoming customers with the highest degree of call satisfaction. This is frequently not the case, because ten years later, the same network infrastructure is still serving the growing network users with little to no maintenance.

The fourth-generation (4G) cellular networks have the reliability and adaptability to multiplex a wide spectrum of traffic from circuit-switched voice to packet-switched voice, data, and multimedia services while retaining the quality of service (QoS) required by service subscribers and their applications '(Kwon & Kim, 2021)). Ajayi *et al* –"(2021) used Nigeria as a case study to assess the efficiency of GSM and CDMA operators and discuss the industry's problems. While GSM stands for global system for mobile communication system, CDMA is Code Division Multiple Access. They came up with strategies for enhancing network efficiency after analyzing the criteria that led to poor service quality by operators. The proposed strategy did not take into account the existence of consumer demands or the ability of the networks. They were primarily concerned with improving the network elements' efficiency.

Vikas & Santosh – (2020) created cellular network performance analysis tools to assess how routing and interconnection charging types impact the link load balance of network. Network initialization, traffic demand generation, simulation of carried traffic demands, and statistical analysis are some of the other network logical units for the simulation software. The logical units are then modeled to use the different routing protocols with the compared results. In this study's context, the simulation is an extension of the current system since it lacks the system's initial state before routing protocols are introduced.

Both users and network providers have been concerned about the continuous rise in the number of wireless mobile users in recent years, as well as the resulting systemic impairments. Unlike the GSM which has a bandwidth-limited network, the universal mobile telecommunication services system (i.e. UMTS), which uses the CDMA system, has an interference-limited network. Owing to inadequate channels and poor radio reception, both band-limited and interference-limited networks have seen an increase in the number of users who are denied access to the network owing to poor radio channel quality and shortage of channels.

Congestion in telecommunications networks is exacerbated by a number of factors, including increased traffic at peak hours and at hot spots. Inadequate network dimensioning, especially the switching system, as well as network operator advertising offers. Congestion is caused by these factors which overload the channel with packets that exceed its capacity. Congestion as a result of irregular public power supply, terminal failure, and other factors are being taken into account, but insufficient network dimensioning, especially the switching and base station system, and transmission failure are not being adequately addressed. Thus, this has necessitated this review.

2. MATERIALS AND METHODS

2.1 MULTIPLE ACCESS NETWORKS AND CONGESTION

Multiple access technique is an implementation of connection between two points through a channel shared by multiple users at different locations. The technique is nearly like that of multiplexing and de-multiplexing, but the multiple access technique is more efficient in radio frequency (RF) than the multiplexing technique because the users are always in different locations and directions. In digital cellular networks, the multiple access allows simultaneous users to share an equivalent channel within a cell. The standard methods are Time Division Multiple Access (TDMA), Frequency Division Multiple Access (FDMA) and Code Division Multiple Access (CDMA).

In TDMA, the allocated bandwidth for the channel is split into time slots; the slots amount depends on the system. Each user is then allocated a slot and this enables multiple users to share an equivalent frequency at different times. FDMA entails the splitting of channel into frequency bands and every user is allocated a waveband. For the CDMA, it enables different users to use same frequency at an equivalent time, but with different spreading code

2.2 Congestion in CDMA

Code Division Multiple Access (CDMA) is a multiple access technique in telecommunication networks that supported spread spectrum technology as propounded by G. A. Shannon. Traffic consideration within the system is given as (1).

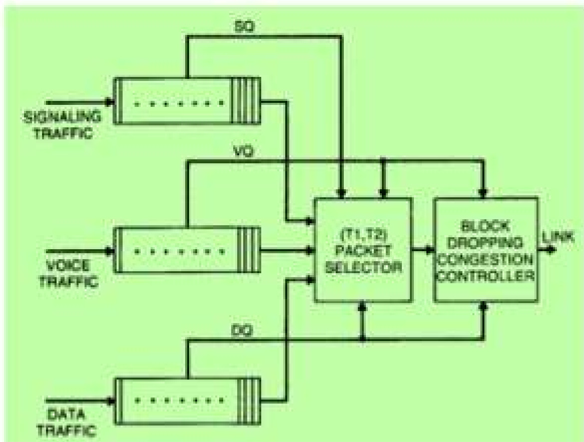

Figure 1. Voice chart of an integrated voice and data multiplexer "(Thew et al., 2015)

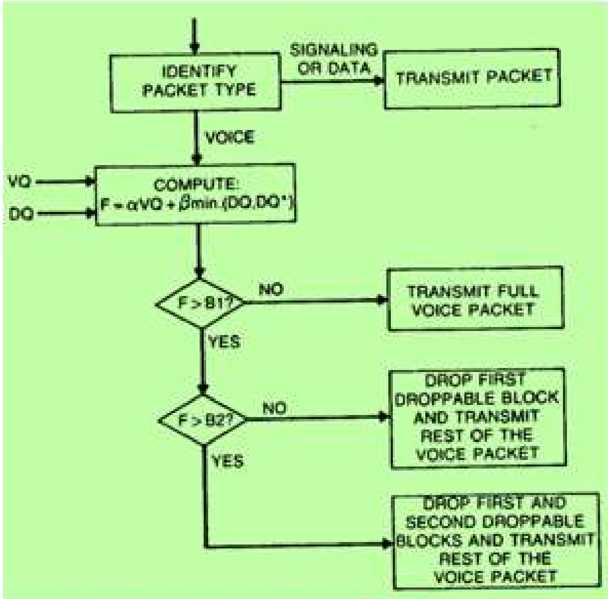

Figure 2. Flow diagram showing operation of block dropping scheme (Thew et al., 2015)

The simulations also accommodate the effect of finite buffers. The frequency at which a voice packet is discarded due to a queue wait time of more than 20ms is used to estimate voice packet loss. According to the findings, the probability of voice packet failure is negligible for a wide range of loads because block dropping on voice packets prevent voice packet loss up to very high loads. Other conclusions drawn from his findings include: voice traffic is well covered under the (T1, T2)-Scheme when data traffic causes overload, while voice quality rapidly degrades under the FIFO-Scheme; voice performance degradation is worse under the FIFO discipline when data traffic is busy; and T1, T2 - scheme offers protection to voice traffic in terms of the mean bits per sample performance at the expense of increased data delay. However, if voice traffic is given top priority over data in order to minimize latency, data traffic can experience unnecessarily long delays and packet losses.

3.2 Optimization Approach to Rate Base Flow Control

Hamadneh et al. (2019) investigated the microscopic behavior of Random Early Marking (REM) for a simple network (1999). REM was compared to the Random Early Drop (RED) and Drop Tail queue management systems in order to provide an optimization approach to rate base flow control. A number of simulations were conducted, each involving four FTP sessions running over TCP Tahoe and data transfer to a common destination. The simulation was used to ensure that the algorithm behaved as predicted by theory.

The source starting time was staggered in the simulation model to allow observation of the Random Early Marking algorithm's actions as the device entered and exited congestion. The sources' utility functions were:

$$a_s \log (1 + x_s) \tag{5}$$

a_s was set to $(C + 1) X \tau$, C is the bottleneck capacity in packets/s, and τ is the session round trip time (4ms). Various step sizes (γ) were used by the router to adjust its link prices in the base case γ was set to 0.1. Fig. 3 shows the topology for the experiment.

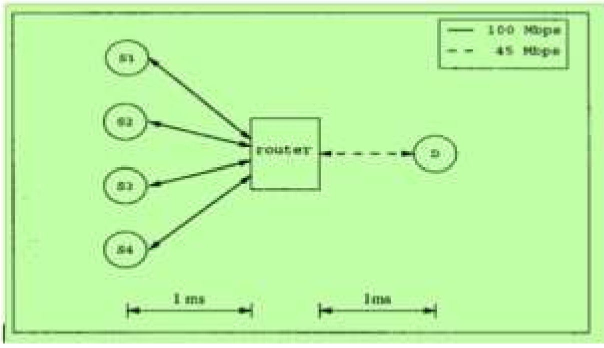

Figure 3. Random Early Marking (REM) Topography (Thew et al., 2015)

According to the findings, the advertised window is very similar to the theoretical equilibrium value of 11.25 for the source advertised window. Each source is also speedily convergent to its fair share value. The three internet Transport Control Protocols REM, RED, and Drop Tail were compared in terms of efficiency in the second experiment. In terms of throughput, the experimental results clearly show that the REM outperforms the RED.

3.2 Congestion Control Based on Accumulation

The internet Transport Control Protocol (TCP) Vegas was generalized by Yong *et. al*. 2005 and He created a general model named ACC Fluid Model by using accumulation and buffered packets flow inside the network routers, as a measure to detect and control network congestion. Using Vegas as a model, a new Monaco scheme was developed that solves the well-known problem of Vegas by using two FIFO priority queues created by two network routers that are congested. It was famous for its backlog estimator's technical problems, which hindered its proper function.

The ACC Fluid Model Network is shown in Fig. 4 where the flow enters into the network at the ingress node R_1 and following it passage through some intermediate nodes R_2 to R_{J-} it goes out through the egress node R_J. The input rate for flow i's at time t in any node is $\lambda_{ij}(t)$ while the output rate is $\mu_{ij}(t)$. Constant value $d_j^2$is the propagation delay from node R_j to node R_{j+1}. Flow i's accumulation as a time-shifted, distributed sum of the queued bits in all the nodes along its path from the ingress node R_1 to the egress node R_J is shown in equation 4.

$$q_i(t) = \sum_{j=1}^{J} q_{ij} \left(t - \sum_{k=j}^{j-1} d_k \, k\right) \qquad (6)$$

$q_{ij}(t)$ is flow i's queued bits in router j at time t.

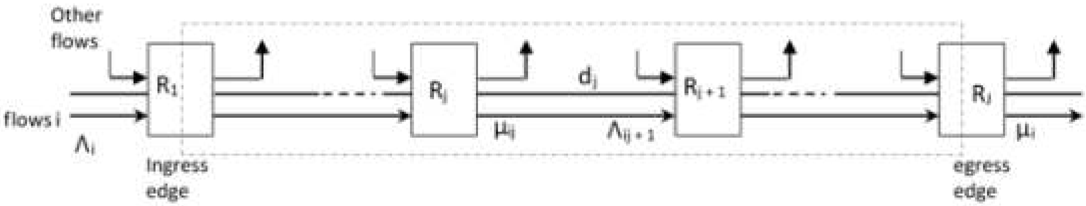

Figure 4. Network Model for ACC (Giordano A. A. & Levesque A. H., 2015).

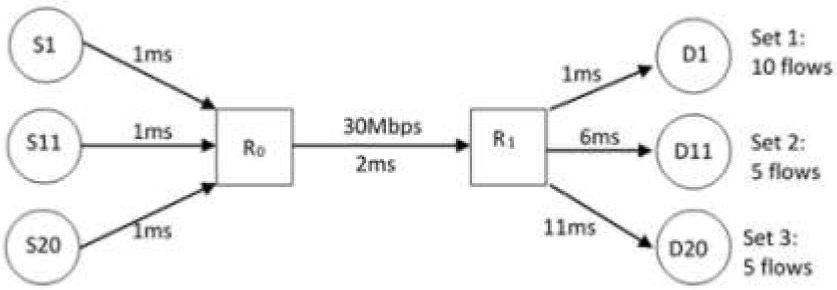

Figure 5. ACC Fluid Model Topography (Giordano A. A. & Levesque A. H., 2015).

In the topology arrangement in Fig. 6, a single 30Mpbs bottleneck with 2ms propagation delay shared by three sets of flows using Monaco scheme is applied. Set 1 starts at 0s and stops at 30s, Set 2 starts at 10s and stops at 40s while Set 3 starts at 20s and stops at 50s (Yong Xia, Harrison, Kalyanaraman, Ramachandran, & Venkatesan, 2005). Although the results show no packets loss, the queue length increases as Set 2 and Set 3 jump in and start decreasing when Set 1 and Set 2 jump out. Also, the throughput decreases as Set 2 and Set 3 jump in and increase as Sets 1 and Set 2 jump out.

Another series of experiments was run using the Monaco Linux scheme implementation to verify the fairness and stability findings from the ns-2 simulations in the previous experiments. One result was obtained from a two-bottleneck topology with complex demands. Two drop tail-bottlenecks, each with 1Mbps bandwidth and a 20ms delay is shown in Fig. 6. Two active short flows such that one long flow going in at 20s and going out at 60s and the other one, long flow active from 40s to 80s were obtained during the 80s experiment. Each flow stabilizes at its proportionally fair after a brief period.

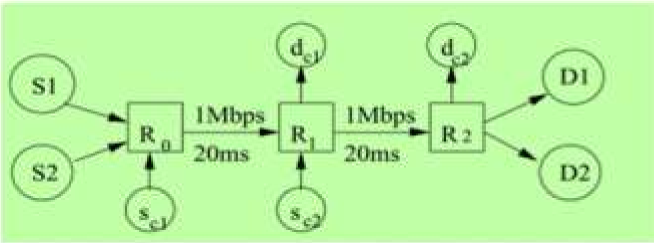

Figure 6 Monaco Linux scheme Topography (Thew *et al.,* 2015)

According to the findings, the throughput for the first long flow starts with 0.33Mbps (its proportionally fair share) and decreases when the second long flow shows up. At the same time, the short flows dropped in its throughput. The second long flow gets about its fair share of 0.33Mbps after 60s. The implementation results oscillate more than the simulation throughput results in previous experiments. This is due to the Linux kernel's restricted timer granularity, which makes traffic less managed than in ns-2.

Given enough buffers in the bottlenecks, the scheme shows its effectiveness in keeping the network secure, equal, and resourcefully utilized. Monaco's output degrades when the buffer is under provisioned, according to their simulations. In order to validate the majority of the simulation realism, it may be necessary to implement Monaco in the Linux kernel based on the click router in order to solve this buffer scalability issue.

3.2 Uncooperative Congestion Control

Fan et al. (2005) proposed an empirical model for re-mapping the utility functions of uncooperative internet flows in order to control them. In the Network Simulator, they introduced the edge-based re-maker (NS). The edge re-maker measured the loss rate for each flow and then re-marked the ACKs using that detail. The Exponential Weighted Moving Average (EWMA) and Weighted Average Loss Indication (WALI) methods of the Equation-based rate control algorithm were used to estimate losses. TCP New Reno's congestion management and loss recovery mechanisms were used in the simulation. The choice for delayed acknowledgements was also disabled. The maximum advertised window was set high enough so that the real window was not obstructed. The model for dealing with Uncooperative Congestion Management is shown in Fig. 8.

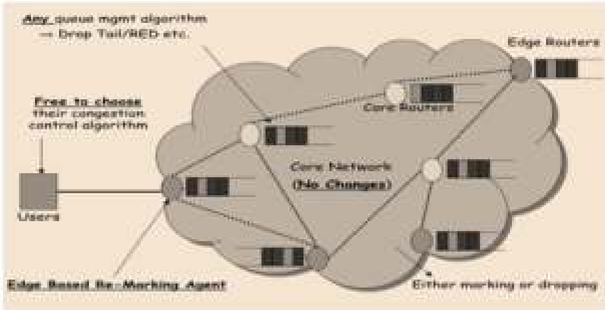

Figure 7. Model for dealing with uncooperative users at network edge –(Theme, 2013)

The relationship to find out the usefulness of any increase/decrease based rate control scheme, U(x), is shown in equations (7) and (8), where *x* represents the rate and R the Round Trip Times RTT.

$$U(x) = \frac{1}{Rxf(x)g(x)} : f(x) \geq 0, 0 \leq g(x) \leq 1 \quad (7)$$

the increase policy *I* and decrease policy *D* are:

$$I : \frac{1}{f(x)} \quad D : g(x) \quad (8)$$

The proposed model can be applied at the network edge, allowing for gradual deployment. Furthermore, the model does not necessitate any changes to the core routers. The model is also independent of the buffer management algorithm, meaning it can be used for both AQM and Drop Tail queues. Finally, even in the presence of high background (web) traffic and reverse path congestion, the model is efficient and works well. Unfortunately, there were no tools for identifying uncooperative users. The purpose of the analysis was not to find uncooperative flows, but to figure out how to handle them.

4 CONCLUSION

Mobile or cellular network uses distributed cell towers that enable mobile devices such as cell phones to automatically switch frequencies and communicate across large geographic areas without interruption. The same basic switching capability enables cellular networks to accommodate many users across a limited number of radio frequencies. Whenever a Base Station (BS) does not have enough space in its queues to put new arriving packets, it is said to be congested. These extra packets will then be misplaced. Due to the congestion, the packets already in the queue take the longest time to be transmitted. Congestion issues cannot be entirely eliminated, but can be reduced by lowering the rate at which the BS alerts the Mobile Station (MS) to transmit data and using the proper buffer administration technique at the BS transmit buffer. Increasing the number of channels per cell also decreases congestion in the BS's receiver and transmit buffers.

A single technique cannot be considered the most basic for curbing congestion. The best strategy for minimizing congestion is determined by the situation, such as the amount of incoming traffic and the rate at which packets arrive. The cost factor is often taken into consideration, as it can have a major impact on the choice of the simplest technique. Finally, the designer must balance what is important against what scenarios are more likely to occur in order to choose an appropriate technique.

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