

Efficient Channel Utilization of Radio Resources in W-CDMA Communication Network Using Dynamic Scheduling Technique

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Abstract

The ever-increasing demand on mobile wireless operators to provide efficient voice and high speed data services has led to search for powerful methods to share the available spectrum in most efficient way. Therefore, in order to guarantee the user QoS requirements with high utilization of the radio resources in Wide Band Code Division Multiple Access (WCDMA)-based cellular networks, both the channel rate and transmission power of each user should be dynamically adjusted. Against this back drop, this work has achieved the incorporation of Dynamic Schedulers (DS) in WCDMA Base stations. The Dynamic Schedulers serve as the main switching technique in the traffic concentration node cells. The Dynamic Schedulers therefore, distribute the available system resources to users in each scheduling time interval according to individual bandwidth and QoS requirement. The work used the MTN RFC 2544 standard for benchmarking the QoS metrics of their WCDMA network in Awka. A validation on the performance of the developed scheme using both field and simulated datasets were obtained. From the assessment of results, the Dynamic Schedulers were able to substantially improve on the performance of the reference network particularly in the areas of dropped calls and latency in all cases. This could lead to improved service delivery in Nigeria as case reference.

Keywords: Dynamic Scheduler, WCDMA, QoS, Throughput, Latency, Call drop, User Equipment.

1. Introduction

The transmission of multimedia-based services over cellular and wireless technologies is on a very steep rise due to the numerous developments in the area of wireless technology over the past decade. This has led to an ever-increasing demand on mobile wireless operators to provide efficient voice and high speed data services. At the same time, the mobile operators support more users per base station in order to reduce overall network cost and make services available to subscribers. Unfortunately, because the available broadcast spectrum is limited, attempts to increase subscriber's density within a fixed bandwidth create more traffic, interference in the system and degrade the signal quality. The need for ever higher spectrum efficiency motivates the search of powerful methods for sharing the available radio spectrum in most efficient way in order to provide communication services with a high capacity and

a good quality of service. In practice, all sharing methods introduce interference which is proportional to the transmitter powers (Riku Jäntti, 2001). This makes resource management in wireless communications a difficult task. One of the 3G wireless networks that is prone to this problem is the Wide Band Code Division Multiple Access (WCDMA). Here, the users spread across both frequency and time in the same channel (Liberti and Rappaport, 1994). Since the multiple users interfere with each other in CDMA systems (Viberbi, 1995), it is crucial to allocate and control the power and transmission rate of each user in order to guarantee the user QoS requirements with high utilization of the radio resource. In the literature, there are two types of CDMA uplink scheduling algorithms: code domain scheduling and time-domain scheduling (Holma and Toskala, 2006). The code-domain scheduling allocates a limited radio resource to many User Equipment (UEs) continuously, inducing guaranteed QoS in terms of delay for real-time traffic. The time-domain scheduling utilizes multi-user diversity to improve system throughput by an opportunistic transmission considering channel status. A combined approach of time and code division scheduling algorithms for WCDMA uplink was presented in (Rosa et al., 2004). The evaluation of three different algorithms that have been utilized – a blind fair throughput scheduler (BFT), a maximum transmit power efficiency scheduler (MTPE) and a channel-state aware fair throughput scheduler (CSAFT) – was assessed through several simulation assumptions, attributing constant throughput and fairness, and capacity increase. The aforementioned algorithms did not take into consideration the additional complexity of the synchronization, the fast variation of the propagation channel and the selection of the transmission power of each Transmission Time Interval (TTI) of the UEs. Hybrid allocation schemes were presented in (You Jin et al., 2007), where the two types of scheduling algorithms were combined exploiting the advantages of each algorithm for best performance. Various traffic services were used, each service to the network was classified and accordingly queued, depending on the service requirement (real-time or not). The proposed scheduling algorithm consists of a traffic classifier, a QoS monitor, a queue selector, and a weight generator. Reasonable enhanced performance was accomplished in terms of packet delay while minimum throughput improvement was achieved.

According to 3GPP standards, the Service Information (SI) does not include any information regarding the delay sensitivity or the current packet delay of each traffic flow. However, this turns out to be a great disadvantage for the scheduling procedure, especially regarding real-time applications, as it does not allow the scheduler to provide delay based QoS differentiation. A modification of the 3GPP specifications in order to include packet delay information in SI was developed by (Dimitrios et al., 2010). Thus, the algorithm achieved the substitution of the Highest priority Logical Channel ID (HLID), with a new metric namely Mean Link Control Protocol (MLCP), which is transmitted on a per-TTI basis from each UE to the Node B. The MLCP criterion was utilized by a Delay-driven Packet Scheduler (DPS) that was designed to provide delay-based QoS. The developed scheme had been evaluated by means of system level simulations. The simulation results showed that the proposed MLCP-DPS scheme achieved significant delay reduction for real-time traffic flows compared to a conventional Class Based Scheduler (CBS).

A code-division generalized processor sharing (CDGPS) fair scheduling scheme that employs both Dynamic Bandwidth Allocation (DBA) and the Generalized Processor Sharing (GPS) discipline to provide fair services in a WCDMA system was proposed by (Liang Xu et al., 2002). The CDGPS scheduler makes use of both the traffic characteristics in the link layer and the adaptivity of the WCDMA physical layer to achieve efficient utilization of radio resources. It adjusts only the channel rate (service rate) of each traffic flow in the WCDMA system by varying the spreading factor and/or using a multiple of orthogonal code channels, rather than allocating service time to each packet. This resulted in lower implementation complexity of the CDGPS scheme than for a conventional GPS-based time scheduling scheme, while the performance was comparable to that of the ideal GPS in terms of bounded delay and guaranteed bandwidth provision.

This work has developed a dynamic scheduling algorithm that distributes the available system resources according to each user's bandwidth requirement. In addition to adjusting the service rate of each traffic flow, it also divides the available spectrum into slots which it assigned to traffic users in each scheduling time interval.

2.0 Material and methods

2.1 Data collection

This work used the MTN RFC 2544 standard for benchmarking the QoS metrics of their WCDMA network in Awka, to ascertain the influence of link adaptation on the WCDMA QoS. The RFC 2544 standard was established by the Internet Engineering Task Force (IETF) standards body, as a method that outlines the tests required to measure and prove performance criteria for carrier Ethernet, WiMax or WCDMA network. Essentially, given the base station setup, the RFC 2544 provides an out-of-service benchmarking method to evaluate the performance of

network using throughput, back-to-back, frame loss and latency tests, with each test validating a specific part of a Service Level Agreement (SLA). These tests follow standard procedures, making it easier for customers to literally observe QoS. The method defines the frame size, test duration and number of test iterations. The RFC describes six out-of-service tests, which means that real traffic must be stopped and specific frames generated to evaluate throughput, latency, frame loss rate, burst tolerance, overload conditions recovery and reset recovery. The observed data presented below, was then used in performance analysis of the developed model.

Table 1: Throughput Test Results

Frame Length (Bytes)	Cfg Rate (Mbps)	Measured Rate (Mbps)	Measured Rate (frms/sec)	Pause Detected
68	45.000	45.000	63920	No
128	45.000	45.000	38007	No
256	45.000	45.001	20381	No
512	45.000	45.003	10574	No
1024	45.000	45.001	5388	No
1280	45.000	45.001	4327	No
1522	45.000	45.002	3648	No

Table 2: Latency (RTD) Test Results

Frame Length (Bytes)	Latency (RTD) (μ s)	Measured Rate (Mbps)	Measured Rate (frms/sec)	Pause Detected
68	2921	45.000	63920	No
128	3035	45.000	38007	No
256	3289	45.001	20381	No
512	3785	45.003	10574	No
1024	4780	45.001	5388	No
1280	5276	45.001	4327	No
1522	5751	45.002	3648	No

2.1.1 System Analysis and Modelling.

In the developed scheme, pair of bandwidth schedulers (Dynamic Bandwidth Schedulers) were assumed to reside in the Radio Network Controller (RNC). The major concern of the Dynamic Schedulers is to minimize pack transmission delay that occurs in a network when the instantaneous arriving rate of a traffic flow exceeds allocated bandwidth. The Dynamic Schedulers also utilized the available network resources efficiently, by ensuring that traffic flow was not allocated more bandwidth than it actually needed.

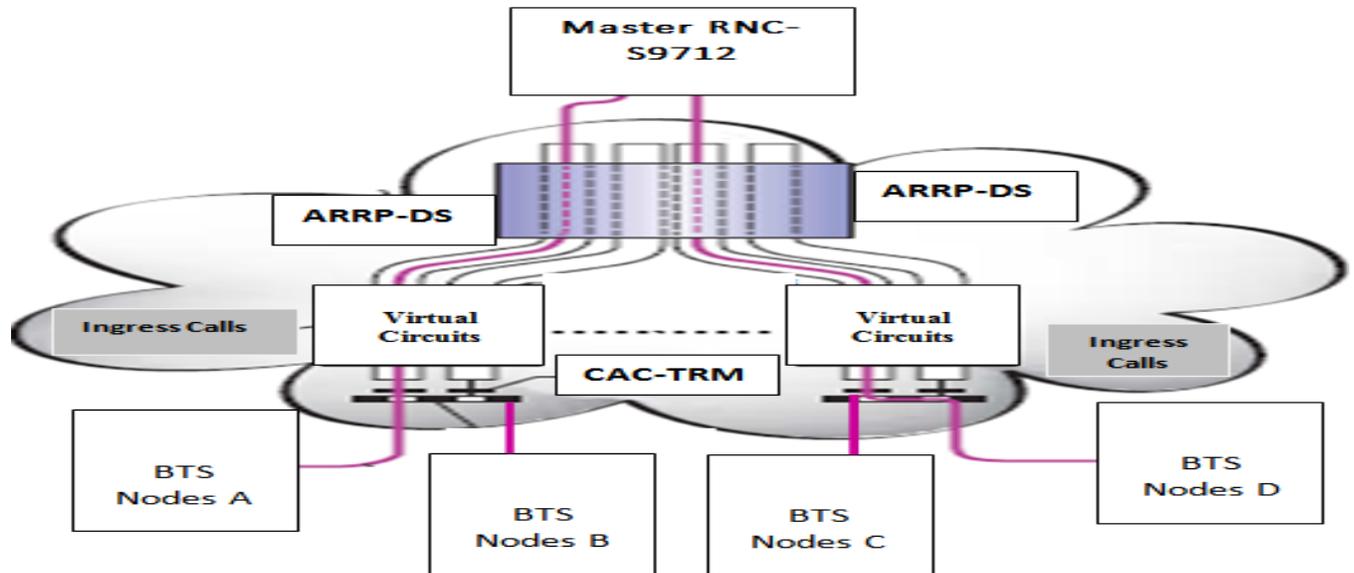


Figure 1: Architecture of switching and termination of Radio Network Controller virtual channel

The Dynamic Schedulers (DS) serve a pure transport aggregating service in the RNC. The DS makes use of Adaptive Resource Reservation Procedure (ARRP) technique to enable several user connections in the radio network layer to be multiplexed flexibly and efficiently on a cascade of traffic concentration node cells. As soon as a user gets admitted into the network, the DS set up virtual circuit connections for the calls. The scheduler allocated a fixed positive real number (called weight) to each traffic flow. The bandwidth was allocated to all flows according to their pre-assigned weights and traffic load. The overall effect was to reduce load in the system and bring the network to equilibrium between priority and non-priority calls. Moreso, the scheduler divides the available channel into different slots which it assigns to active users at each transmission time interval T_i . When an admitted user wants to send a packet in the next time interval, T_{next} , the user may send the transmission requests to the base station in the current time interval, T_{cur} . Upon receiving bandwidth request from all active users, the base station classifies the request based on their class taking into account user QoS requirements and available uplink capacity; and stores into a FIFO type class queue (buffer). The resource allocation message in terms of the spreading factor and the number of codes of each user will be sent via downlink channels before the start of the next time slot. The base station executes the proposed scheduling algorithm at the end of the time interval. Due to the burstiness of call packet traffic, sometimes a user may not have packets to transmit and gives up its bandwidth for a while (Liang Xu et al 2002), the excess bandwidth can be distributed among all backlogged sessions at each instant in proportion to their individual weights.

2.1.1.1 Dynamic Call Scheduling Algorithm

To achieve maximum or effective throughput, dynamic scheduling is used to increase spatial resource reuse and fair access to the RNC. For scheduling calls, the DS determines if the call is a priority or less/non priority one by checking its status using the defined power rating and transmission rate indexing. It further checked the virtual circuits using ARRP and verified the remaining slots on the service frames to ensure that it gives QoS for next call ingress or admitted calls. Algorithm 1 describes this process while Algorithm 2 monitors the QoS requirement.

Algorithm 1: Dynamic Scheduling

Input: Users in BTS , Arrival rates , Call duration , Priority id , Buffer Size , Virtual Circuit connection V_{cc} , Power index P_r , Transmission rate Index T_m ;

Output: Admitted calls ;

Begin ()

For (Power index $P_r = 1$ && Transmission rate Index $T_m = 1$);

While Time (Call duration) uplink duration

All subscribers with call packets transmits (Set=1);

DS checks for power enabled calls and verifies their transmission dates;

V_{cc} link connection // establishes link connection;

Ds feedback estimated empty buffers-RNC, to BTS,

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BTS arrival time;
Call packets Cps stored in DS// respective buffers;
ARRP terminates statistical multiplexing using  $P_r$  &  $T_{rm}$ ;
End While
While time downlink duration;
If call scheduling index buffer > 0 Then
Case_1: switch high priority calls;
Case_2: switch best effort calls;
Else
Schedule Non-real time calls;
End if;
End While;
End for;
End

```

Algorithm 2: QoS Monitoring

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Require:// This algorithm takes new connection request (Req) as input and
// allocate RNC slot if QoS test are satisfied (admitted)
i = re.src; j = req.dest; R = Find Common RNC parent (i,j)//
Check_QoS(req, Hs, Hp, end-end) // connect admitted source to RNC
Check_QoS (req, Hp, Ht)// RNC sink to external sources/targets
Accept the request and allocate slots;
End if;
// ChecQoS (req, start_hop, end_hop);
Vcc_no_of_slots = req.reqBW;
For i = Start_hop to End_hop Do
If no_of_slots > slots [i] Then;
Reject the request (req) and return;
End if;
Else;
If no_of_slots < slots [i] Then;
Admit the request and return;
End if
End

```

This algorithm ensures that the QoS at the DS is not compromised

2.1.1.2. Flowchart of the Dynamic Scheduling Algorithm

The flowchart of the Scheduling algorithm is shown in fig.2. From the flowchart, it is assumed that Scalable Orthogonal Frequency Division Multiplexing (SOFDMA) takes care of multipath effects, Doppler shift, and pathloss challenges. It actually increases the number of subcarriers leading to better immunity to inter-symbol interference (ISI). Beside the power and transmission rate indexing enforced by the scheduler, it is assumed that SOFDMA serves as an automatic modulation that adjusts the system so as not to affect real time calls at event of unfavourable network conditions. The virtual circuit connection ensures that calls that pass the entire QoS test are admitted; else absolute call rejection is enforced.

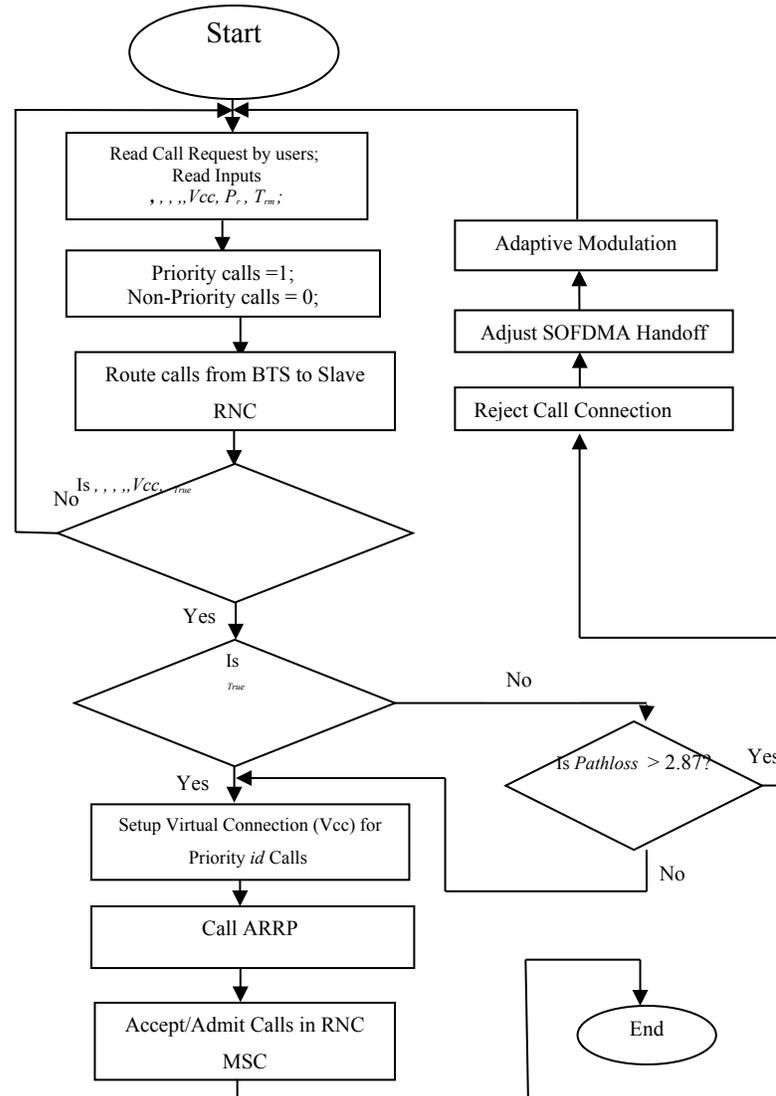


Figure 2: Flowchart of the Rate Scheduling Algorithm

2.1.1.3 SIMULATION MODEL

To achieve the test case network model, a widely acceptable WCDMA simulator known as Riverbed modeler _ae_175A_PL6_13310_win was deployed. Testbed setup in Modeler was run by representing WCDMA real world devices as nodes and links. Modeler provides an environment on which attributes of these nodes and links can be configured and used as inputs in the simulation run, after which results are analysed. Each project can contain at least one scenario. A scenario can then be edited in the project editor where subnets, nodes, links, utilities, and application traffic can be included for the simulation study. The Discrete Event Simulation (DES) takes all the configurations and embedded algorithms in place into account and models the behaviour. During the process, the selected statistics are collected and saved for viewing after the run. A simple process characterization of 173 developed WCDMA system is illustrated in Fig.3.

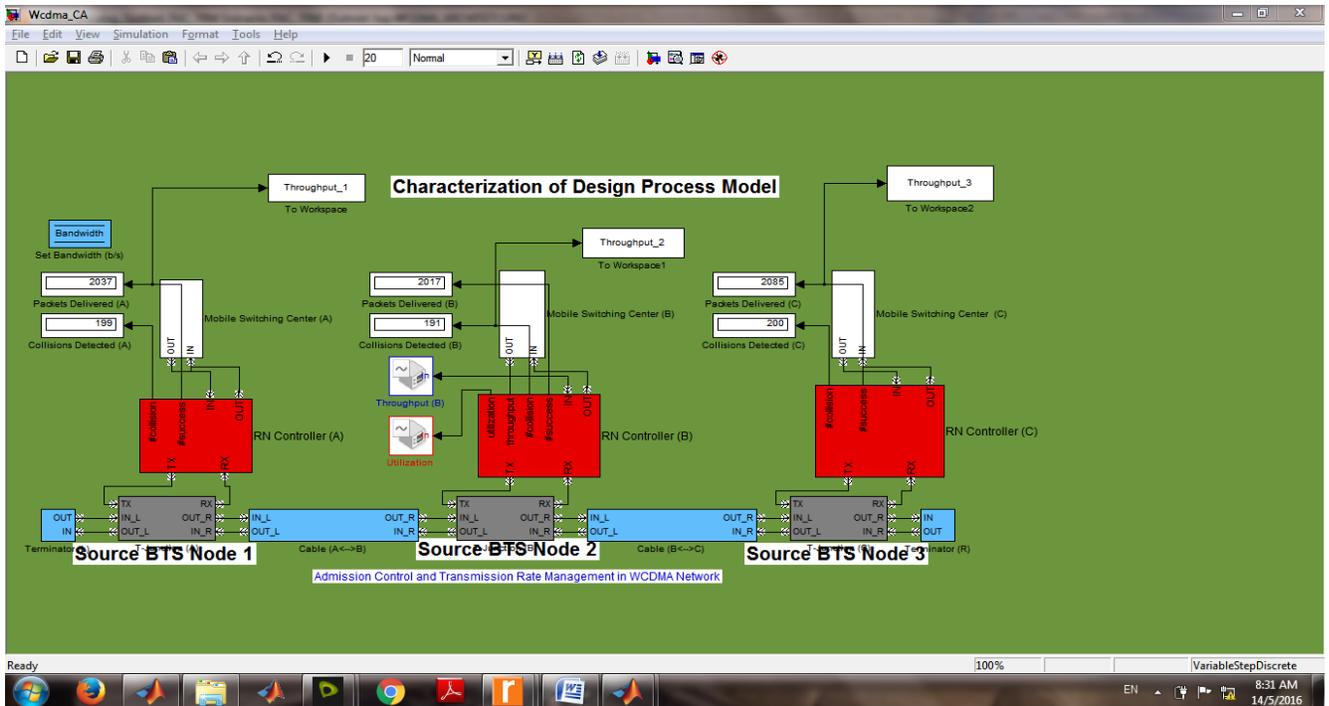


Figure 3: Process model WCDMA system

3.0 Results and Discussions

3.1 Throughput

It is the amount of traffic (real time or non-real time) that is received in the destination, represented in Megabits/second. Fig.4 compares the throughput obtained with the dynamic scheduling algorithm against the measured throughput from MTN network. It can be seen that the proposed dynamic scheduling technique has better throughput compared with the MTN network throughput. In the plot, the developed algorithm throughput gradually increased when the number of users was increased. We observed that MTN network had 45Mbps (3.93%) as its loaded throughput while the developed dynamic scheduling technique offered 1100Mbps (96.01%) throughput. The reason for better throughput is due to the very adaptive nature of power control that allows the network to offer quality of service for different call traffics by efficiently adjusting the power and improving the transmission capacity of the network. Hence, the user voice calls or data calls were transmitted successfully.

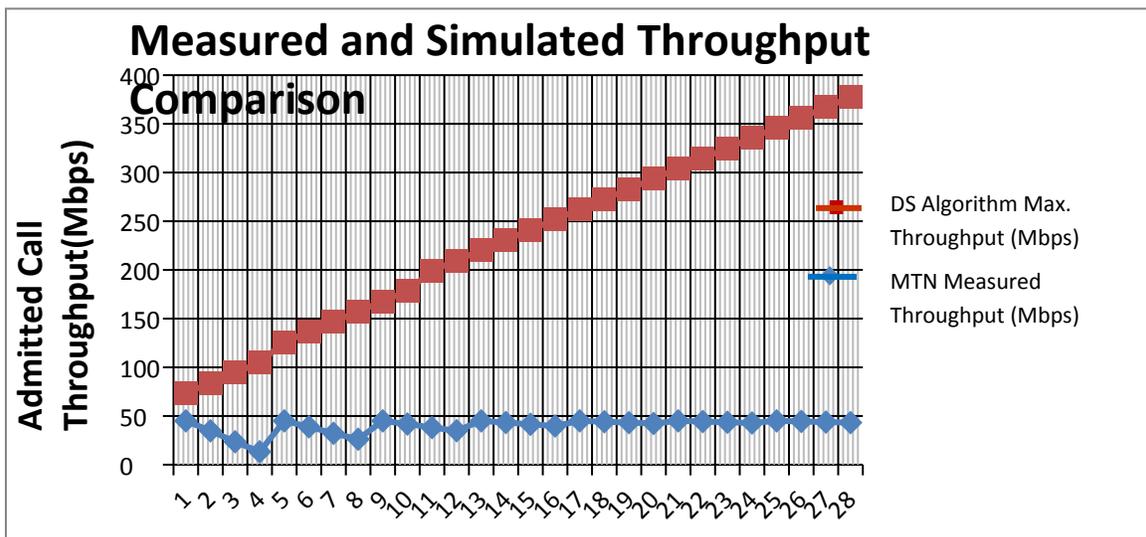


Figure 4: User Frame length throughput comparison

3.2 Latency

It is the average end to end delay that occurred at the destination for all flows. Fig.5 shows the comparative plots. In the developed system, a latency of 0.001205sec (16.72%) was observed while a maximum latency of 0.006sec (83.28%) was observed for MTN network. A low latency plot results from enhanced throughput behaviour.

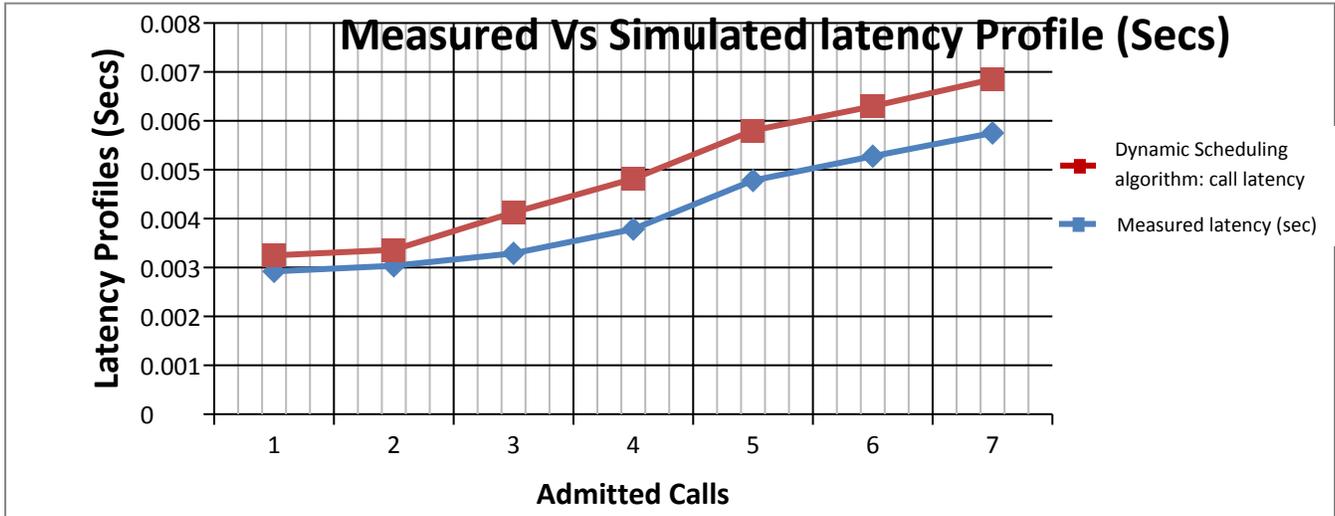


Figure 5: End to End call latency comparison

Fig.6 shows the admitted call load plot from the virtual circuit. Usually, once calls are admitted by the slave RNC, the transmission rate determined by the dynamic scheduler is optimized because the bandwidth priority for the real time flows is scheduled. For every call that is not rejected, transmission priority is guaranteed for its flows. The VCC load observed in this case is about 252,000 bits/sec. This implies that the dynamic scheduler for rate management and the power control for the calls must determine the acceptable throughput that the load profile will offer at all times.

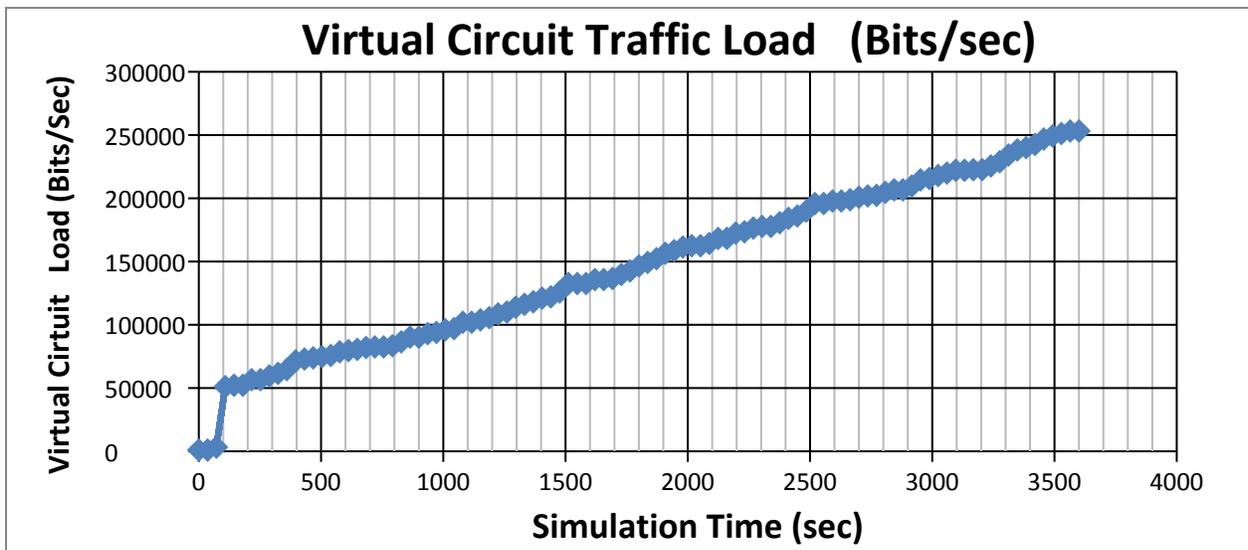


Figure 6: Virtual circuit connection load trend

4.0. Conclusion

From the assessment of results, the Dynamic Schedulers were able to substantially improve on the performance of the reference network particularly in the areas of dropped calls and latency in all cases. The number of overloaded cells could be at least halved and the interference was reduced by adequate tilting while achieving lower latency and optimal throughput. Hardly any useful throughput improvement was achieved with the existing MTN RFC snapshot model. The explanation for the comparatively weak results in the existing networks is the type of admission control strategy employed. Based on these developments, there is a variety of questions to pursue in the future. First of all, a careful assessment of the estimation errors of a typical average-based load calculation is needed. A good matrix design optimization model can be developed for quantifying the throughput metric.

5.0 Recommendation

The model developed in this work was able to achieve better throughput performance and reduced call drop compared to MTN network. Existing network providers in Nigeria should therefore explore the practicability of AM-CAC in their network so as to remain compatible with future scalable WCDMA networks. However, the use of adaptive neural network for load balancing in radio access networks should be investigated and explored. Also, the use of RFC 2544 benchmarking for out-of-service tests viz: throughput, latency, frame loss rate, burst tolerance, overload conditions recovery and reset recovery should be restructured to serve as Key Performance Indicator (KPI) metrics for WCDMA networks in Nigeria.

References

Arad M.A and Leon-Garcia A., "A Generalized Processor Sharing Approach to Time Scheduling in Hybrid CDMA/TDMA," *Proc. IEEE INFOCOM '98, San Francisco, CA, Mar. 1998, pp. 1164–71.*

Dimitrios KOMNAKOS, Dimitrios SKOUTAS, Demosthenes VOUYIOUKAS, Angelos ROUSKAS, "Scheduling Optimization for Real-Time Services in WCDMA", *Future Network & Mobile Summit 2010 Conference Proceedings*
Paul Cunningham and Miriam Cunningham (Eds) IIMC International Information Management Corporation, 2010 ISBN: 978-1-905824-16-8.

Harri Holma and Antti Toskala, "HSDPA/HSUPA for UMTS" John Wiley & Sons, 2006.

Liang Xu, Xuemin Shen, and Jon W. Mark, "Dynamic Bandwidth Allocation with fair scheduling for Wcdma Systems," *IEEE Wireless Communications*. April 2002.

Liberti J.C. Jr. and Rappaport T.S. (1994), "Analytical results for capacity improvements in CDMA", *IEEE Transactions on Vehicular Technology*, vol.43, no.3, pp.680-690.

Riku Jäntti(2001), "Power control and transmission rate management in cellular radio systems", Helsinki University of Technology Control Engineering Laboratory Espoo may 2001. Report 123.

Rosa C., Outes J., Sorensen T.B., Wigard J., Mogensen P.E., "Combined time and code division scheduling for Enhanced Uplink packet access in WCDMA", *IEEE 60th Vehicular Technology Conference, (VTC2004-Fall)*, Vol. 2, pp. 851-855, September 2004.

You Jin Kang, Junsu Kim, Dan Keun Sung, and Seunghyun Lee, "Hybrid Scheduling Algorithm for Guaranteeing QoS of Real-Time Traffic in High Speed Uplink Packet Access (HSUPA)" *Proc. of IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC'07)*, pp.1-5, September 2007.

Viterbi A.J, *CDMA: Principles of Spread Spectrum Communication*.
 Boston, MA: Addison-Wesley, 1995.