Design and Simulation of Voice and Critical Data apriority Queue (VCDPQ) Scheduler for Constrained-Bandwidth VoIP Networks

J. N. Dike, C. I. Ani

Abstract — VoIP technology entails the transmission of digitized voice conversations across a packet-switched network such as the Internet. It promises to integrate the data and voice networks into a single network. However, there remain some very significant quality of service (QoS) impairments (such as latency, packet loss, echo and jitter) surrounding this technology, which degrade voice quality. The original Internet architecture was designed for best effort (non real time) services and does not guarantee QoS for real-time applications. Voice and business/mission-critical data (B/MCD) are typical examples of real-time traffics. This work addresses the effect of delay on the perceived voice and B/MCD traffic quality. It presents the design and simulation of voice and critical data priority queue (CCDPQ) scheduler for constrained-bandwidth VoIP networks. The scheduler incorporates mechanisms to achieve a graceful tradeoff between priority and fairness to all traffics as a solution to the transmission impairments in the evolving growing VoIP networks. The proposed algorithm defining the different levels of abstraction was analyzed. Riverbed (OPNET) Modeler was used to simulate the performance of the proposed scheme. The results obtained in the simulation show that the hybrid architecture achieves better latency and resource utilization than similar existing schedulers.

Index Terms — Business critical data, constrained bandwidth, delay, mission critical data, quality of service, resource utilization, voice.

1 INTRODUCTION

VoIP or Internet Telephony entails the transmission of digitized voice conversations (or traffic) across a packet-switched (IP) network such as the Internet. It promises to integrate the data and voice networks into a single network. VoIP is one of the most important technologies in the world of communications today and is the most important service in the broadband network with growing attempts focused on improving voice quality to match that in PSTN [1]. Increasingly, the expectation placed upon VoIP networks is that they will provide the same or better voice quality than traditional telephone, thereby assuring Quality of Service (QoS) for voice and business/mission critical traffics over the network [2].

However, there remain some very significant QoS impairments (such as latency, packet loss and jitter) surrounding this technology, which degrade voice quality [3]. The original Internet architecture was designed for best effort services, whereby packets can be delivered out of order, corrupted, duplicated or not at all. Moreover, packets take different amount of time to reach their destinations [4]. The Internet therefore does not guarantee QoS for real-time applications [5].

In the quest to optimize the QoS of VoIP networks, many schemes have been proposed to address the needs of both delay-sensitive (real-time) and best effort (non real-time) traffic flows. Some approaches aim at preventing congestion by limiting load and using priority scheduling [6]. In all, low latency, low jitter and low packet loss to the streaming flows as well as fair resource sharing are guaranteed only if the rate of the streaming flow is relatively a very small fraction of the link rate [7], [8], [9]. However, with the expected increase in the volume of voice traffic in the network as the traditional telephone system gradually but progressively migrates to VoIP, the use of a fair scheduler will severely degrade voice quality. On the other hand, the use of the priority scheme will lead to a significant unfair bandwidth sharing.

Again, there is not much research work done in offering same special preference to business/mission critical data (B/MCD) traffic (such as real-time online purchases, security alerts, bank transfers, weather forecasts, remote/emergency environmental monitoring, disaster alerts, military commands, remote industrial control systems, and so on) as accorded to voice traffic. In other words, earlier solutions to the QoS challenges of VoIP networks have been focused on giving precedence to voice traffic at the expense of B/MCD traffic.

Critical in the Nigerian Telecommunication Industry is the lack of functional broadband access networks [10]. This work therefore addresses the effect of delay on the perceived voice and B/MCD traffic quality in constrained-bandwidth VoIP networks. The dominant causes of delay in packet networks are fixed propagation delays on wide

- *J. N. Dike is currently pursuing a PhD degree program in the Department of Electronic Engineering, University of Nigeria, Nsukka, Nigeria, PH: +234(0)8052764438. E-mail: justus.dike@uniport.edu.ng*
- *C. I. Ani is a Professor of Communication Engineering in the Department of Electronic Engineering, University of Nigeria, Nsukka,, Nigeria, E-mail: cosmas.ani@unn.edu.ng*

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area links and variable queuing delays in switches and routers. Since propagation delays are a fixed property of the network topology [11], overall delay and jitter are minimized when the variable queuing delays are minimized. Furthermore, if queues remain short relative to the buffer space available, packet loss is also kept to a minimum.The hybrid optimized QoS scheme proposed in this work models an approach of adaptively evaluating and policing incoming IP flows as well as classifying and mapping different traffic types for individual applications or users. It incorporates mechanisms to achieve a graceful tradeoff between priority and fairness to all traffics as a solution to the transmission impairments in the evolving growing VoIP networks. The developed algorithm defining the different levels of abstraction was analyzed. Riverbed (OPNET) Modeler was used to simulate the performance of the optimized architecture. An algorithmic analysis of the scheduler has been published in [2]. To implement the architecture, an Internet Service Provider's (ISP's) network is configured accordingly. The designed model is implemented in two stages, namely: for wired and wireless network topologies. In this paper however, only the wired network topology is presented.

2. RELATED WORKS

Related works in optimizing the QoS of a VoIP network have been focused on traffic-scheduling algorithms to ensure either minimum traffic delay constraints or fair resource sharing to all applications running on the network [2]. Several queuing strategies have been proposed as solution to support the delay-constrained traffics in a best-effort network [12], [13], [14], [15], [16]. In all these proposals however, no precedence was given to B/MCD traffic flows.

A QoS-guaranteed network normally differentiates between different types of traffic and provides different treatments to the traffics. This is made possible by using either the type-of-service (ToS) [17] bits or the differentiated services (DiffServ) [18] [19] [20], [21], [22], [23] field in the IP header, or still through the use of signalling protocols such as: resource reservation protocol (RSVP) [24], [25] and multi-protocol label switching (MPLS) [26]. Traffic identification can also be implemented by configuring network devices to support prioritization based on physical port, protocol, IP address, transport address or packet length [27].

3. METHODOLOGY

The design and analysis of the proposed scheduler architecture and algorithm are hereby presented. Riverbed (OPNET) Modeler was used to validate and simulate the scheduler in a constrained-bandwidth VoIP network. Results obtained by evaluating the network performance using delay and resource utilization metrics were then discussed.

3.1 Overview of the Proposed Scheduler Architecture

The proposed optimized QoS architecture [2] is an integration of several technologies. It is comprised of the Packet Classifier, the Token Bucket, the Differentiated Services (DiffServ) and the Weighted Round Robin (WRR) Scheduler modules. The hybrid architecture is illustrated in Figure 1.

The Packet Classifier module consists of two packet classifiers. Classifier1 is used to classify the packets of the incoming source traffic (p) into two main classes, namely: voice ($p1$) and non-voice ($p2$) flows [7], [28]. Packet Classifier2 is used to classify the non-voice flows into three other classes, namely: business/mission-critical data (B/MCD, ($p3$)); video ($p4$) and best-effort data ($p5$) traffics. The essence of Classifier2 is to capture and accord business/mission-critical data flows (such as on-line purchases, bank transfers, weather forecasts, disaster alerts, and so on) the necessary priority and fairness they deserve.

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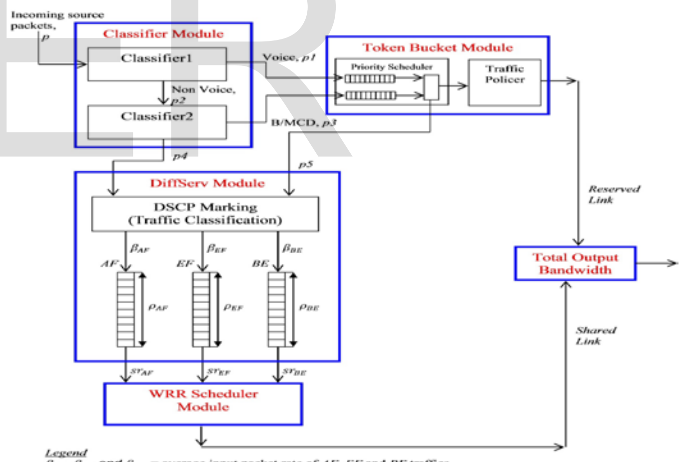

Legend
β_{AF}, β_{EF} and β_{BE} = average input packet rate of *AF*, *EF* and *BE* traffics.
ρ_{AF}, ρ_{EF} and ρ_{BE} = buffer size (or average queue length) for *AF*, *EF* and *BE* traffics.
sr_{AF}, sr_{EF} and sr_{BE} = service rate (or queue weight) for *AF*, *EF* and *BE* traffics.

Figure 1: The optimized hybrid scheduler architecture

The dynamics of network traffic flow and packet distinguishing is implemented by the input service routine architecture, which applies traffic congestion avoidance controls [29], [30] to the incoming flows and places incoming packets into separate queues for subsequent processing by inspecting the type-of-service (TOS) [17] bits in the packet IP header. Non-preemptive priority scheduling discipline is employed for forwarding voice and business/mission-critical data (B/MCD) traffics to the Token Bucket. This implies that there is no interruption to any traffic being transmitted through the Bucket. Voice traffic is classified into the high priority class while B/MCD traffic is classified into the low priority class at the output queue.

The Token Bucket module is used to split the incoming voice or business/mission-critical data traffic into two sub-flows [7]. The first sub-flow is a well shaped flow with maximum rate equal to γ bits/second generated by the Token Bucket. The second sub-flow is the packet (*p6* - still of voice or business/mission-critical data traffic) rejected by the Token Bucket.

In the DiffServ module, video traffic is mapped to Assured Forwarding (AF) traffic class. Voice or business/mission-critical data traffic, which was rejected from the Token Bucket is mapped to the Expedited Forwarding (EF) traffic class. The remaining data traffic (such as email, file transfer, and so on) is mapped by default to the Best Effort (BE) class.

The WRR scheduler module is used to adaptively regulate the bandwidth utilization among the competitive traffic flows from the DiffServ module. The output (constrained) bandwidth is divided into two parts, namely: the reserved (dedicated) link and the shared link. The reserved link is used to service the specified portion of voice or business/mission-critical data traffic from the Token Bucket. The shared link is used to service the other traffics as scheduled fairly and adaptively by the WRR scheduler.

3.2 An Algorithmic Analysis of the Proposed Scheduler

The pseudo code of the proposed scheduling algorithm is presented in Table 1. Using the top-down design approach, the following algorithm defining the various activities performed at every level of abstraction (module) is analyzed.

3.2.1 The Packet Classifier Module: The incoming source packet (p) is classified (by Classifier1) as it arrives the edge network device into two classes, namely: voice ($p1$) and non-voice ($p2$) (lines 1-2). If the packet is voice ($p1$), it is directed to the Token Bucket module and placed in the high priority queue (lines 3-4), otherwise it is directed to packet Classifer2 (lines 5-6). If the non-voice packet is business/mission critical data (B/MCD ($p3$)), it is directed also to the Token Bucket module and placed in the low priority queue (lines7-8), otherwise it is directed to the DiffServ module (lines 9-10).

3.2.2 The Token Bucket Module: The voice ($p1$) and B/MCD ($p3$) traffic flows are directed to the Token Bucket module by executing the non-preemptive priority scheduling discipline (lines 11-12). Each traffic flow is divided into two parts: the first part (the reserved voice or B/MCD flow) is well shaped with a maximum rate (generated by the Token Bucket) equal to $\boldsymbol{\gamma}\, bits/second$. The second part comprises the voice or B/MCD flow that is rejected from the Token Bucket module because of overflowing the rate γ (line 15). The reserved voice or B/MCD flow is served directly through a reserved link dedicated only for this purpose (lines 13-14).

3.2.3 The DiffServ Module: The surplus voice or B/MCD ($p6$) flow (rejected from the Token Bucket) is redirected to the DiffServ module (line 16) where it is marked and classified as EF traffic (lines 38-40). Also, if the non-voice flow from Classifier2 is video ($p4$), it is marked and classified as AF traffic (lines 17-20), otherwise it ($p5$), is marked and classified by default as BE traffic (lines 28-30). The three classes of traffic are respectively observed, assigned logical queues, the buffer size (length or load) of each queue is calculated and the queues directed to the WRR scheduler module (lines 21-27, 31-37 and 41-47).

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Table 1: Scheduling control algorithm of the optimized model

QoS Scheduling Algorithm

Packet Classifier

1. **On the arrival of IP packet** (p) to the scheduler
2. Class = Classifier1 ($p1$):
3. **If** (class = voice ($p1$)) **then**
4. Send to the Token Bucket via the high priority queue;
5. **Else**
6. Send to (Classifier 2 ($p2$));
7. **If** (class = business/mission critical data (B/MCD), ($p3$)) **then**
8. Send to the Token Bucket via the low priority queue;
9. **Else**
10. Send to (DiffServ ($p4$) and ($p5$));

Token Bucket

11. **On arrival of IP packets** ($p1$) and ($p3$) to the Token Bucket
12. Execute the non-preemptive priority scheduling discipline: $Voice = HIGH$, $B/MCD = LOW$;
13. **If** (size ($p1$) or ($p3$)) <= current token) **then**
14. Send to the reserved link;
15. **Else**
16. Send to (DiffServ ($p6$));

DiffServ

17. **On arrival of IP packets** ($p4$) and ($p5$) to the DiffServ module
18. **If** (traffic = video ($p4$)) **then**
19. Mark with AF DSCP value; **then**
20. Classify as AF traffic; **then**
21. Observe input packet rate: β_{AF};
22. Observe encoding and packetizing delay: δ_{AF};
23. Observe number of sessions: N_{AF};
24. Observe average packet size: PS_{AF}; **then**
25. Assign logical queue to traffic; **then**
26. Calculate buffer size: $\rho_{AF} = \frac{\beta_{AF} * \delta_{AF} * N_{AF}}{PS_{AF}}$; **then**
27. Send to (WRR scheduler);
28. **Else**
29. Mark ($p5$) by default as BE DSCP value; **then**
30. Classify as BE traffic; **then**
31. Observe input packet rate: β_{BE};
32. Observe encoding and packetizing delay: δ_{BE};
33. Observe number of sessions: N_{BE};
34. Observe average packet size: PS_{BE}; **then**
35. Assign logical queue to traffic; **then**
36. Calculate buffer size: $\rho_{BE} = \frac{\beta_{BE} * \delta_{BE} * N_{BE}}{PS_{BE}}$; **then**
37. Send to (WRR scheduler);
38. **On arrival of IP packet** ($p6$) to the DiffServ module
39. Mark with EF DSCP value; **then**
40. Classify as EF traffic; **then**
41. Observe input packet rate: β_{EF};
42. Observe encoding and packetizing delay: δ_{EF};
43. Observe number of sessions: N_{EF};
44. Observe average packet size: PS_{EF}; **then**
45. Assign logical queue to traffic; **then**
46. Calculate buffer size $\rho_{EF} = \frac{\beta_{EF} * \delta_{EF} * N_{EF}}{PS_{EF}}$; **then**
47. Send to (WRR scheduler);

WRR Scheduler

48. Observe the current buffer sizes: ρ_{AF}, ρ_{BE} and ρ_{EF};
49. Observe traffic priority values: $PV_{AF} = 2$, $PV_{BE} = 1$ and $PV_{EF} = 3$; **then**
50. Compute service rates: $sr_{AF} = \frac{\rho_{AF}(t) * PV_{AF}}{\rho_{EF}(t) + \rho_{AF}(t) + \rho_{BE}(t)}$

$$sr_{BE} = \frac{\rho_{BE}(t) * PV_{BE}}{\rho_{EF}(t) + \rho_{AF}(t) + \rho_{BE}(t)}$$

$$sr_{EF} = \frac{\rho_{EF}(t) * PV_{EF}}{\rho_{EF}(t) + \rho_{AF}(t) + \rho_{BE}(t)}; \textbf{ then}$$

51. **If** Max_{SR} = maximum of (sr_{AF} or sr_{BE} or sr_{EF}); **then**
52. Send to shared link;
53. **Else**
54. **If** $sr_{AF} = sr_{BE} = sr_{EF}$; **then**
55. Send to shared link using priority value;
56. **Return** Max_{SR}.

3.2.4 The WRR Scheduler Module: The buffer size or current offered load (queue length) and priority of each of the traffic classes are observed (lines 48-49). The observed parameters are used to compute the service rate (or queue weight) of each of the traffic classes (line 50). The traffic class that has the maximum weight is serviced first, then the next, and so on (lines 51-52). If the computed weights are equal to each other, the traffic class that has the highest priority value is serviced first, then the next, and so on for every round-robin session (lines 53-55). This process is repeated for as long as there are available traffic queues for transmission (line 56).

4 SIMULATION TOPOLOGY

The simulation of the voice and critical data priority queue (CCDPQ) scheduler for constrained-bandwidth VoIP networks is implemented using Riverbed (OPNET) Modeler. Riverbed Technology Inc is a leader in Application Performance Infrastructure that delivers the most complete platform for Location-Independent Computing. Riverbed Modeler provides a modelling and simulation environment for designing communication protocols and network equipment. It models and analyses the behavior of the entire network, including its routers, switches, protocols and servers as well as predicts the performance of IT infrastructures including individual applications and networking technologies [31].

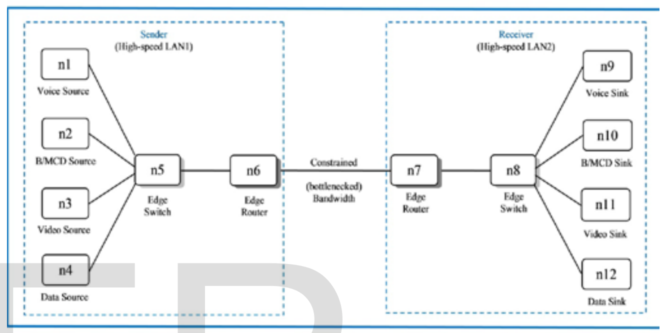

Figure 2: Simulation Network Topology

The simulation network topology is shown in Figure 2. Traffics generated from source nodes n1, n2, n3 and n4 were respectively classified as voice, business/mission-critical data (B/MCD), video and best-effort data. These nodes are connected to the edge switch (n5), which in turn is connected to the edge router (n6), all at the sending end. The edge router is then connected to the network via a constrained-bandwidth link (2Mbps). The connection is similar but reversed at the receiving end. The edge router (n7) connects the edge switch (n8), which in turn connects the voice (n9), B/MCD (n10), video (n11) and best-effort data (n12) sinks. The marking of packets is performed by the edge switch (n5) while scheduling of packets through the network is performed by the edge router (n6). The local area networks (LAN1 and LAN2) at the sender and receiver ends are high-speed LANs. The link between the source nodes and switch is 100Mbps while that between the switch and router is 1Gbps. In this paper, the simulation focused on packet end-to-end or mouth-to-ear (M2E) delay and resource utilization performance metrics of the proposed voice and critical data priority queue (VCDPQ) scheduler.

5 RESULTS AND DISCUSSION

The simulation results are presented and discussed in this section. Figure 3 shows the end-to-end (or mouth-to-

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ear) delay performance evaluation of the proposed algorithm compared with those of Class-Based Weighted Fair Queue (CBWFQ), Application-Aware Scheduler (App-AS), Low-Latency Queuing (LLQ), Contention-Aware Temporary Fair Scheduling (CATFS) algorithm and Low Latency and Efficient Packet Scheduling (LLEPS) algorithms for a single simulation run. The plots show that the proposed QoS model guarantees better quality assurance in terms of packet delay.

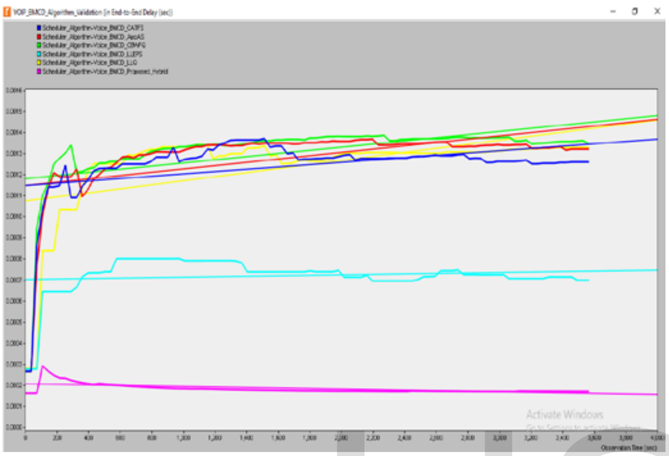

Figure 3: End-to-End Delay Performance Evaluation of Proposed Algorithm compared with CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms.

CBWFQ is an extended WFQ. It does not guarantee low-delay requirements for real-time applications such as voice. Algorithms like LLQ uses strict priority queues to provide minimum but policed bandwidth as well as low-delay guarantee to real-time applications. In terms of delay, LLEPS offers better performance than LLQ, App-AS and CATFS especially in streaming applications. But the proposed hybrid combines these optimization scheduling qualities and allows much robust flexibilities for user-defined classes, queue management and bandwidth guarantees especially for voice and B/MCD traffic flows. From the delay metrics in Figure 4, the Proposed, CATFS, App-AS, CBWFQ, LLEPS and LLQ algorithms respectively have 3.13%, 21.09%, 22.65%, 21.88%, 10.94%, and 20.31% delays. This shows that the proposed scheduler offers better traffic provisioning in constrained-bandwidth networks considering the incremental source traffic intensities.

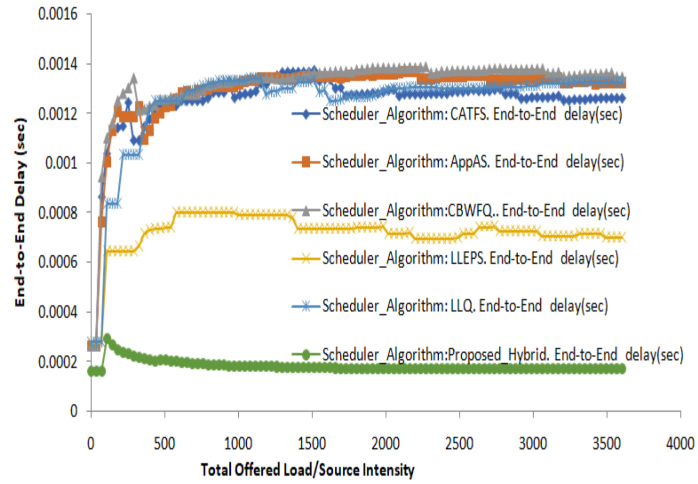

Figure 4: Validation plot of End-to-End Delay against Total Source Intensity for the Proposed, CBWFQ, App-AS, LLQ, CATFS and LLEPS Algorithms.

Figure 5 shows the resource utilization performance evaluation of the proposed algorithm compared with those of CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms for a single simulation run. Resource Utilization accounts for the fair utilization of the available (constrained) bandwidth by all traffics in the network. The plots also show that the proposed QoS model guarantees better resource utilization, thereby ensuring optimized network performance.

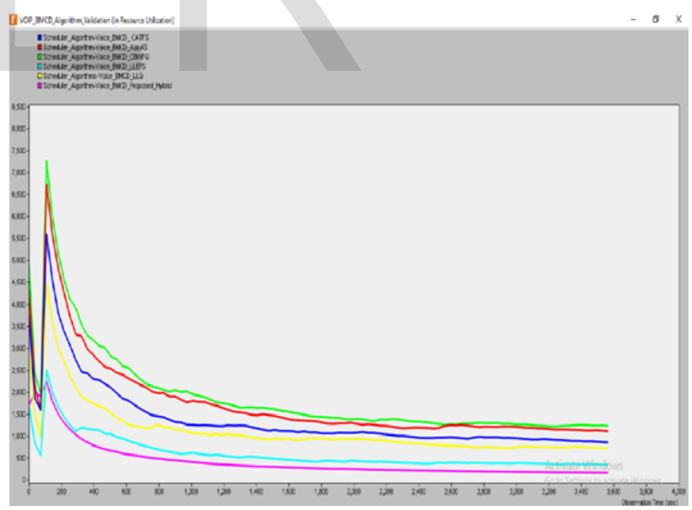

Figure 5: Resource Utilization Performance Evaluation of Proposed Algorithm compared with CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms

In Figure 6, the comparison of resource utilization plotted against total source load intensity for the proposed algorithm with CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms is shown. From the utilization metrics generated from the Riverbed modeler, the respective algorithms had pre-fixed multiplier factor (10^{-3}) for the individual datasets. Now, by resolving the multiplier, as the offered load increases, resource utilization exponentially increased and then returned to its steady state. The Proposed hybrid,

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CATFS, AppAs, CBWFQ, LLEPS, and LLQ algorithms had utilization ratio of 0.05%, 23.90%, 21.73%, 27.16%, 10.86% and 16.30% respectively. This means that with the proposed algorithm, the offered load will have the least impact on the constrained bandwidth. The network datasets generated from the simulation runs for this plot are given in the Appendix (Table A13). This is also better that the WFQ and WRR scenarios previously discussed.

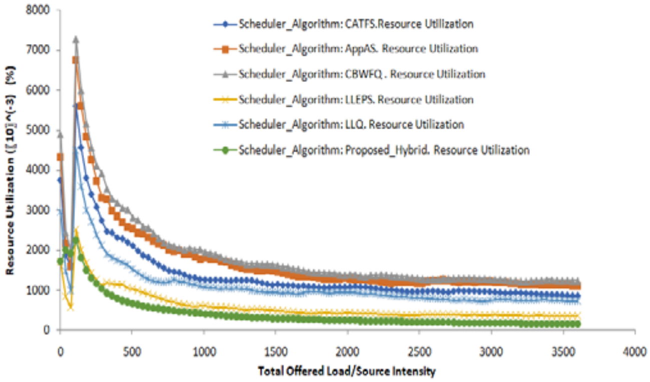

Figure 6: Validation plot of Resource Utilization against Total Source Intensity for the Proposed, CBWFQ, App-AS, LLQ, CATFS and LLEPS Algorithms.

6 CONCLUSIONS

By the built-in mechanisms, the proposed scheme ensures that adequate precedence is given to both voice and B/MCD traffics even with the expected increase in voice traffic workload on the network. The proposed scheme also ensures that adequate fairness in resource allocation and utilization is maintained. The optimized performance of the proposed scheme therefore guarantees a graceful tradeoff between priority (to voice and B/MCD traffics) and fairness (to all network traffics) in a constrained- bandwidth VoIP network without over provisioning the users. Implementing the proposed model therefore ensures that: (i) every packet arriving the network is classified and explicitly marked for proper identification; (ii) adequate precedence is given to both voice and business/mission critical data traffics; (iii) the provision of a dedicated (reserved) link handles the demands of the expected rapid increase in voice traffic; (iv) excess voice and business/mission-critical data packets that should have been lost are recovered and serviced with due priority; (v) other traffics (video and best-effort data) in the network are given fair treatment in the allocation of available resources, and (vi) fairness in resource sharing ensures that queues do not grow excessively, thereby reducing the delay and packet loss impairments. The results obtained in the simulation show that the hybrid architecture achieves better latency and resource utilization than similar existing schedulers. The design has been structured to be simple and easy to understand. It is developed in a modular form for easy manipulation. The structure also makes the scheduler architecture robust and consistent in its operation.

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