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Validation of Critical Data and Voice Hybrid Scheduler (CDVHS) for Cyber-Physical Computer Systems (CPCS) in Constrained-Bandwidth VoIP Networks

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Abstract

Consequent upon advances in networking technology is the emergence of converged networks, enabling a new generation of integrated data and voice application. Voice over Internet Protocol (VoIP) networks are converged multiservice systems in which various class types (CTs) broadly classified as real time and non-real time, share the converged transmission and management infrastructure as well as other resources of the network. Real-time traffics are delay-sensitive. Cyber-physical computer systems (CPCS) are new generation of systems which find application in industrial/process control and automation systems. Information generated by CPCS is generally classified as critical data and are therefore delay-sensitive. The volume of data generated by these emerging systems will continue to increase the load on the converged network. This work therefore proposes an optimized hybrid scheduling architecture that evaluates, polices, classifies and maps incoming IP flows into different class types. Using packet loss probability and end-to-end packet delay quality of service metrics, the work evaluates the performance when high volumes of business/mission-critical data (B/MCD) and voice traffics are transmitted in constrained-bandwidth VoIP networks. Critical data is captured, separated from best-effort data and given due precedence/priority. A structured algorithm defining the different levels of abstraction was developed and described. Riverbed Modeler, version 17.5 was used to validate and simulate the performance of the hybrid scheduler.Comparison with those of Class-Based Weighted Fair Queue (CBWFQ), Application-Aware Scheduler (App-AS), Low Latency Queuing (LLQ), Contention-Aware Temporary Fair Scheduling (CATFS) and Low Latency and Efficient Packet Scheduling (LLEPS) algorithms respectively gave packet loss evaluation of 0.83%, 22.71%, 26.50%, 18.93%, 24.22% and 6.81%. Delay evaluationrespectively gave3.13%, 21.09%, 22.65%, 21.88%, 10.94%, and 20.31%. These results show that the hybrid architecture achieves betterpacket-loss probability and latency than similar existing schedulers. Comparison of the impact of different coding schemes on the performance of the proposed model respectively gave packet loss evaluation of 60%, 33.33% and 6.67%for G.711, G.726 and G.729.Delay evaluation respectively gave34.19%, 33.86% and 31.94%. These results show that the G.729 coding scheme offers lower packet loss and better end-to-end delay at 8kbps bit rate and therefore guarantees better network QoS performance.

Keywords: *Critical data, hybrid scheduler, cyber-physical computer systems, quality of service, structured algorithm, constrained bandwidth.*

1. Introduction

Consequent upon advances in networking technology is the emergence of converged networks that has enabled a new generation of integrated data and voice applications.Cyber-physical computer systems (CPCS) are new generation of systems with integrated computational, networking

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and physical capabilities that can interact with humans through many new modalities [1]. These systems find application in industrial and process control systems, cloud robotics and automation as well as in medical devices and systems among others. Information generated by CPCS is generally classified as critical data and are therefore delay-sensitive. The volume of critical data generated by these emerging CPCS will continue to increase [2, 3]. However, the original Internet architecture was designed to transport best-effort traffic and therefore would not guarantee quality of service (QoS) for delay- (or time-) sensitive traffics over wired and wireless networks. So, one of the underlying reasons for QoS into the Internet is the desire to provide high-quality support for delay-sensitive traffics such as all critical data (classified in this work as business/mission-critical data (B/MCD)), Internet Protocol (IP) voice and video services. When the traffic load level is low, the network delivers a high quality of service. As the load levels increase, the network congestion levels increase, and service-quality levels decline uniformly, [4]. Thus, it is expected that network administrators should be able to ensure their networks meet the stringent requirements for QoS of B/MCD and voice traffics especially in terms of end-to-end packet delay and packet loss.

The focus of most converged network strategies is Voice over Internet Protocol (VoIP) [5, 6]. VoIP networks are multiservice systems in which various class types (CTs) share the transmission, connection/transport switching, management, and other resources of the network. Class types are broadly classified as real time and non real time, which include various groupings and aggregations of services, such as B/MCD, voice, video and best-effort data. Business/mission critical data includeremote industrial/process control systems and automation, electronic voting systems, remote medical monitoring systems,real-time online purchases, security alerts, bank transfers, weather forecasts, remote/emergency environmental monitoring, disaster alerts, military commands, and so on. The interesting reality is that converged networks promise to deliver lower operating costs and easier service deployment [7]. Hence, optimization of the QoS obtained in VoIP networks has become one of the industry's research priorities. Again, only a hybrid scheme can effectively solve the transmission impairment factors of the converged networks [8, 9].

This work presents the validation an optimized critical dataand voice hybrid scheduler (CDVHS) for cyber-physical computer systems in constrained-bandwidth VoIP networks. A number of factors contribute to constraints or bottlenecks in the available bandwidth. These constraints impede optimal network performance. In the Nigerian Telecommunication Industry for example, the poor state of functional broadband access networksis primarily responsible [10, 11, 12].The work therefore investigatespacket loss probability and end-to-end packet delay performance when high volumes of business/mission-critical data and voice traffics are transmitted inconstrained-bandwidth VoIP networks. The impact of different coding schemes on the performance of the proposed hybrid architecture was also investigated. Hence, comparative analyses ofpacket loss probability and end-to-enddelayQoS performance parameters were carried out with G.711 (64kbps), G.726 (32kbps) and G.729 (8kbps) coding schemes. The architecture evaluates, polices, classifies and maps incoming IP flows into different class types. Critical data is captured, separated from best-effort data and given due precedence/priority. The proposal incorporates adequate congestion control mechanisms to accommodate all applications running in the network. Using the top-down design approach, a structured algorithm defining the different levels of abstraction was developed and analyzed. Riverbed Modeler, version 17.5 [13] was used to validate and simulate the performance of the hybrid scheduler.Packet loss probabilityand end-to-end packet delay QoS metrics are used to evaluate the performance of the proposed model. The proposed architecture is hereby implemented on a wired-network topology [9].

2. Earlier Proposals

Earlier optimizationproposals [14, 15, 16, 17, 18, 19, 20, 21, 22] of the QoS of VoIP networks havefocused on granting higher preference/priority to voice traffics and lumping all data traffic flows together. However, giving strict preference to voice traffic flows in an IP network imposes serious constraints on the fair utilization of the available bandwidth to other contending flows, especially to business/mission-critical data (B/MCD). The growing evolution of CPCS, whereby embedded computers and networks monitor and control physical processes, has brought with it the need for frameworks, algorithms, methods and tools to satisfy the high reliability and security requirements for the heterogeneous cooperating components in the system [1]. Due to the criticality of CPCS, they are often required to be high-confidence [2]. Again, in these days of threatening global economic recession and socio-political insecurity, electronic military commands, industrial/process remote control, electronic businesses/services as well as distress/emergency alerts arising from re-occurring global natural disasters, the need to separate B/MCD from best-effort data flows has become imperative, especially in constrained-bandwidth networks [9]. In Nigeria for example, the public demand for the introduction of electronic voting machines in the reformed electoral system further justifies the separation of such critical data from best-effort data.

3. Proposed Hybrid Scheduler Architecture

The proposed hybrid scheduler architecture [9, 23, 24, 25] illustrated in Figure 1 comprises the Packet Classifier, the Token Bucket, the Differentiated Services (DiffServ) and the Weighted Round Robin (WRR) Scheduler modules [16, 26, 27, 19, 8]. 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 (*p*1) and non-voice (*p*2) flows [16, 28]. Packet Classifier2 is used to classify the non-voice flows into two other classes, namely: business/mission-critical data (B/MCD, *p*3) and others (*p*4 - consisting of video and remaining (best effort) data traffics). The essence of Classifier2 is to capture and accord all business/mission-critical data traffics the necessary precedence/priority they deserve.

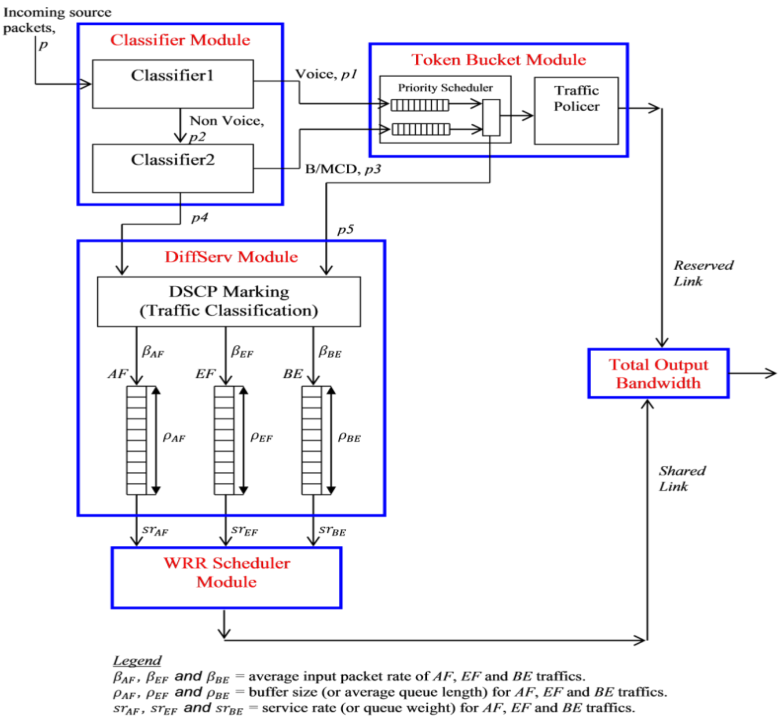

Figure 1: The proposed optimized hybrid scheduler architecture

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) [31] 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 [16]. 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.

4. Developed Algorithm for the Proposed Scheduler

The top-down design approach was used to develop a structured packet scheduling algorithm that defines the various activities performed at every level of abstraction (module). This is presented in the signal flowchart [32] of Figure 2.

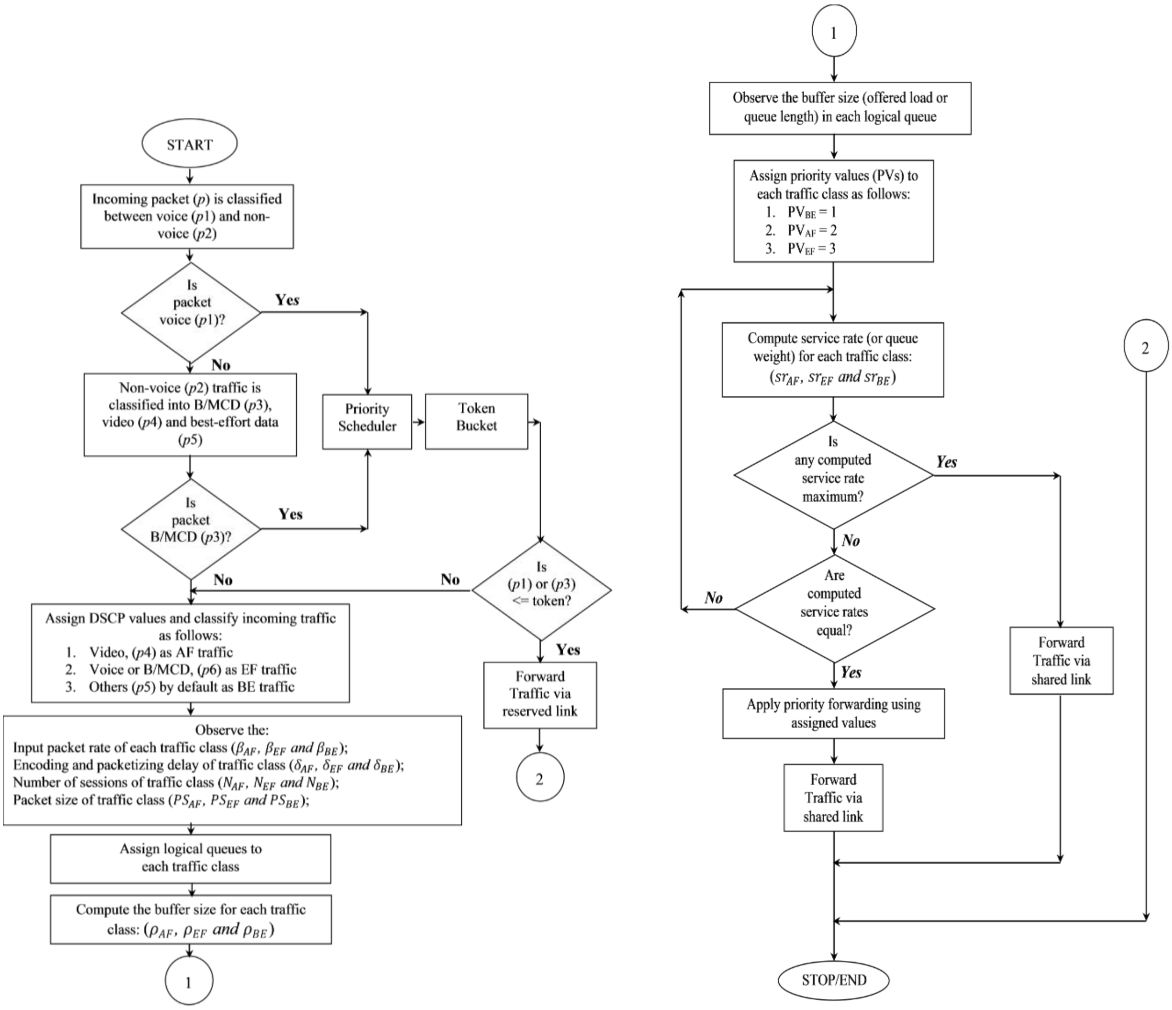

Figure 2: The structured signal flowchart of thedeveloped scheduler

5. Validation and Simulation of the Hybrid Model

The proposed QoS-based model is hereby validated and simulated with Riverbed Modeler, version 17.5 [13]. Figure 3 [9] illustrates the network topology used for the validation and simulation. Voice, B/MCD, video and data source (generator) nodes were respectively configured with type of service (ToS) 1, 2, 3 and 4. These nodes are connected to the edge switch (N5), which in turn is connected to the edge router (N6), all at the sender end. The edge router is then connected to the network via a constrained (bottlenecked) bandwidth. The connection is similar but reversed at the receiver end. The edge router (N7) connects the edge switch (N8), which in turn connects the voice, B/MCD, video and best-effort data sinks.

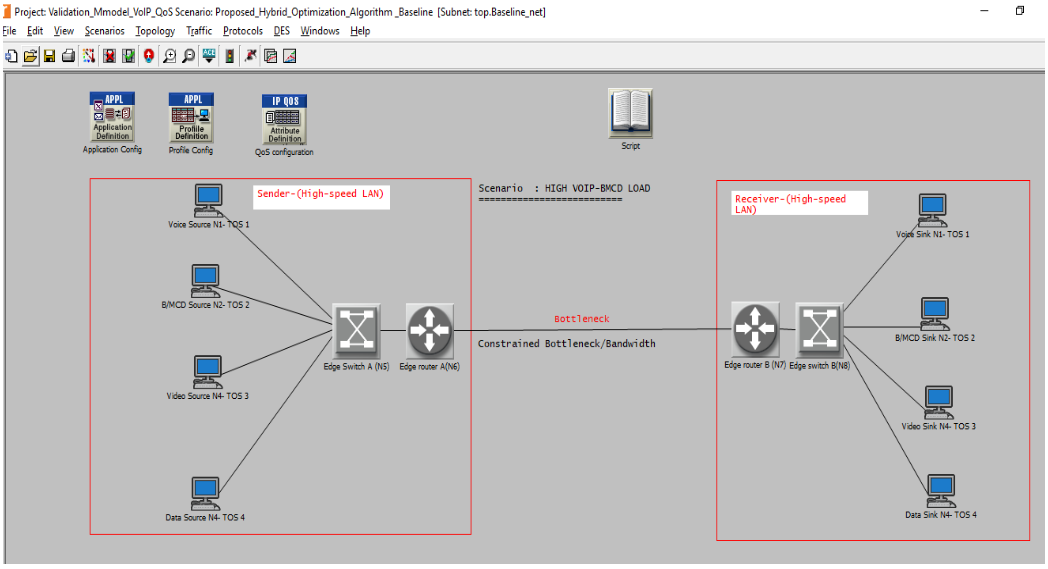

Figure 3: The simulation network topology.

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 link capacity between the source nodes and edge switch A (N5) is 100Mbps while that between N5 and N6 (edge router) is 1Gbps. The capacity of the bottlenecked-link is 2Mbps. Using end-to-end packet delay and packet loss probability quality of service (QoS) metrics, the performance of the proposed hybrid architecture was compared with those of the Class-Based Weighted Fair Queue (CBWFQ) algorithm; Application-Aware Scheduler (App-AS) algorithm; Low Latency Queuing (LLQ) algorithm; Contention-Aware Temporary Fair Scheduling (CATFS) algorithm and Low Latency and Efficient Packet Scheduling (LLEPS) algorithm.

6. Simulation Results and Discussions

The results obtained from the simulation runs are hereby presented and discussed. The results ofpacket-loss probability and latency (or end-to-end packet delay) performance metrics of the proposed optimized hybrid architecture were compared with those of Class-Based Weighted Fair Queue (CBWFQ) algorithm; Application-Aware Scheduler (App-AS) algorithm; Low Latency Queuing (LLQ) algorithm; Contention-Aware Temporary Fair Scheduling (CATFS) algorithm and Low Latency and Efficient Packet Scheduling (LLEPS) algorithm.

Figure 4 [9] shows the packet loss probability performance evaluation of the proposed algorithm compared with those of CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms. The packet loss probability is plotted against time, when 70% (of total source intensity) input data rates of voice and B/MCD traffics are transmitted with 30% (of total source intensity)input data rates of video and best-effort data over the network during one simulation run from one domain in the network. The Figure shows that the proposed hybrid QoS-based model guarantees lower packet loss probability thereby ensuring optimized network performance.

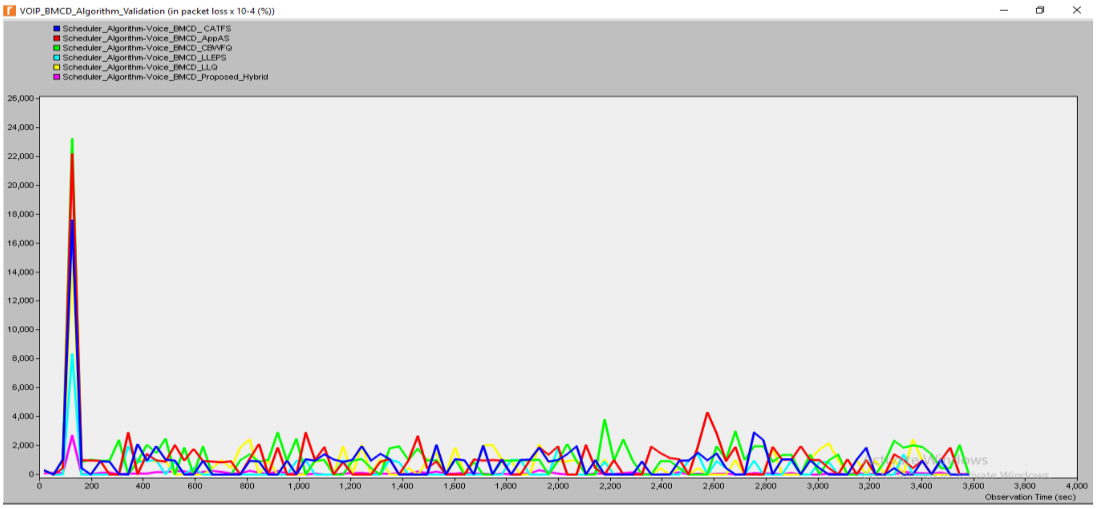

Figure 4: Packet loss probability performance evaluation of proposed algorithm compared with CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms

Figure 5 [9] shows the comparison of packet loss probability plotted against total source load intensity for the proposed algorithm with CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms. From the packet loss metrics generated from the Riverbed modeler, the respective algorithms had pre-fixed multiplier factor (10^{-4}) for the individual datasets. Now, by resolving the multiplier, the Proposed, CATFS, App-As, CBWFQ, LLEPS and LLQ algorithms respectively have packet losses under the incremental or varied source load intensities as 0.83%, 24.22%, 26.50%, 22.71%, 6.81%, and 18.93%. This behavior shows that both the proposed algorithm and LLEPS have the most reliable packet loss statistics with the proposed having the least packet loss under the constrained-bandwidth scenario.

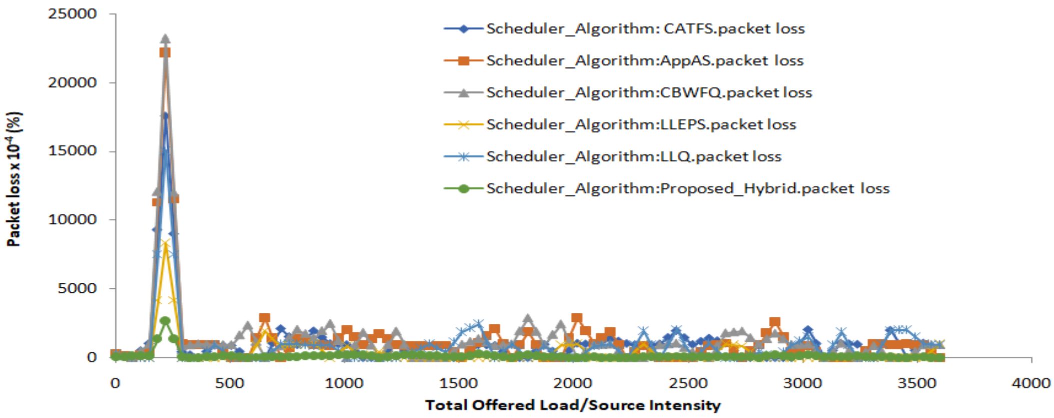

Figure 5: Validation plot of packet loss probability against total source intensity for the proposed, CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms

Figure 6[9, 23] shows the end-to-end packet delay performance evaluation of the proposed algorithm compared with those of CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms, when 70% (of total source intensity) input data rates of B/MCD and voice traffics are transmitted with 30% (of total source intensity)input data rates of video and best-effort data over the network during one simulation run from one domain in the network.The Figure shows that the proposed QoS model guarantees better quality assurance in terms of packet delay.

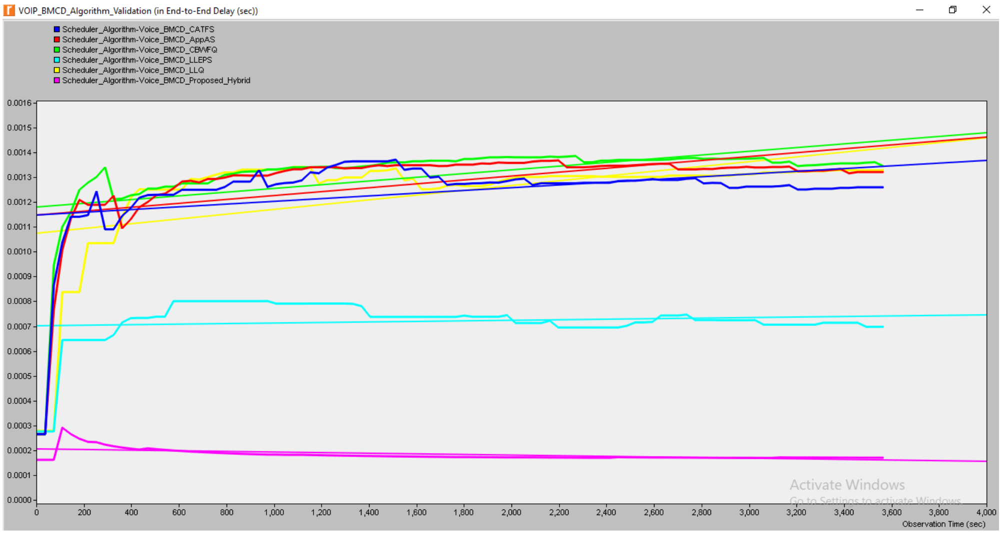

Figure 6: End-to-end packet delay performance evaluation of proposed algorithm compared withCBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms

Figure 7[9, 23] shows the comparison of end-to-end packet delay plotted against total source load intensity for the proposed algorithm with CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms. From the delay metrics, 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.

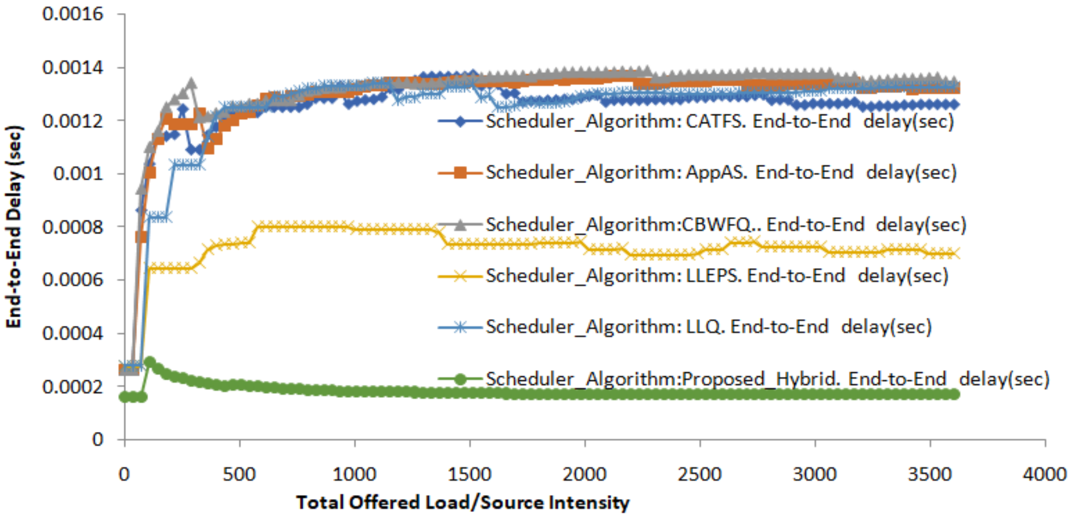

Figure 7: Validation plot of end-to-end packet delay against total source intensity for the Proposed, CBWFQ, App-AS, LLQ, CATFS and LLEPS algorithms

Figure 8shows a comparative analysis of the impact of the coding schemes on the performance of the proposed model in terms of packet loss probability. From the Riverbed statistic engine script, the plot of G.711, 726, and 729 offered 60%, 33.33% and 6.67% respectively. The Figure shows that G.729 guarantees lower packet loss in the network, thereby validating the optimal performance of the proposed QoS-based hybrid model.

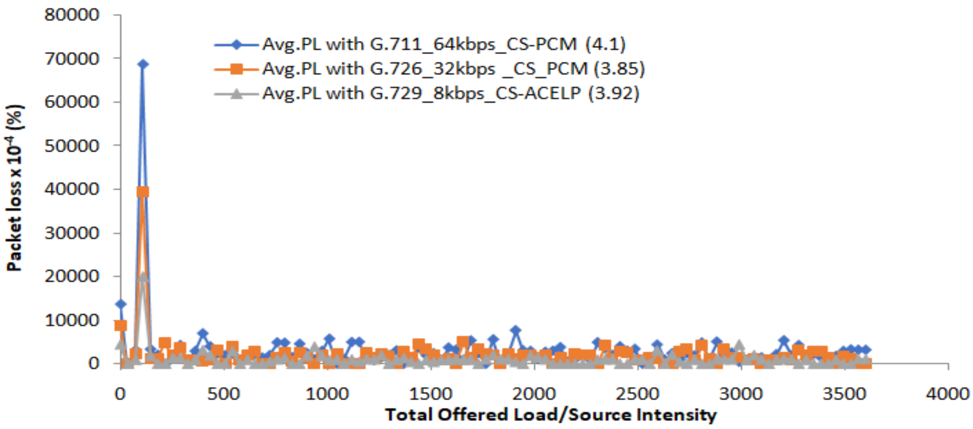

Figure 8: Average packet loss probability evaluation using G.711, G.726 and G.729 coding schemes

Figure 9 shows a comparative analysis of the impact of the coding schemes on the performance of the proposed hybrid model in terms of end-to-end packet delay. In the network, the variation of the source intensities gave 34.19%, 33.86% and 31.94% respectively for G.711, G.726 and G.729. The plot shows that the G.729 coding scheme offers the best end-to-end delay at 8kbps bit rate and therefore guarantees better network QoS performance. Recall that of the three coding schemes investigated, the G.729 has the least bit rate. This fact further validates the optimal performance of the proposed QoS model.

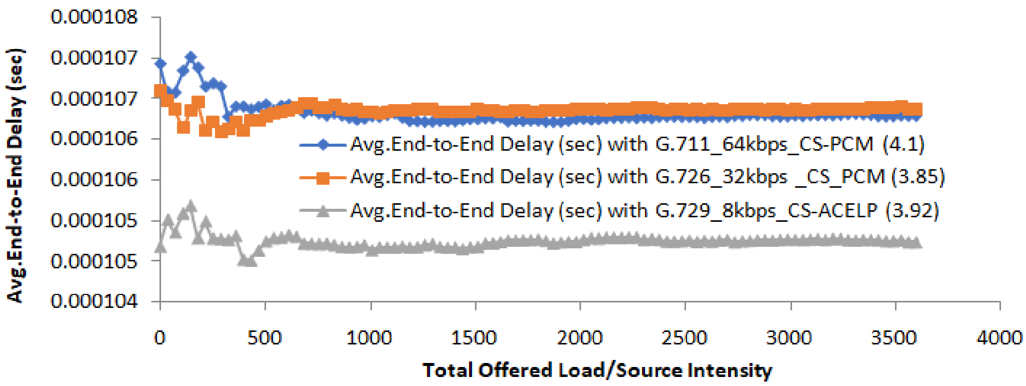

Figure 9: Average end-to-end delay evaluation using G.711, G.726 and G.729 coding schemes

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CONCLUSIONS

In this paper, the validation of an optimized QoS-based critical dataand voice hybrid scheduler for cyber-physical computer systems in constrained-bandwidth VoIP networks has been presented. Packet loss probability andend-to-end packet delay quality of service metrics were used to evaluate the performance of the proposed algorithm. Comparison with those of Class-Based Weighted Fair Queue (CBWFQ), Application-Aware Scheduler (App-AS), Low Latency Queuing (LLQ), Contention-Aware Temporary Fair Scheduling (CATFS) and Low Latency and Efficient Packet Scheduling (LLEPS) algorithms respectively gave packet loss evaluation of 0.83%, 22.71%, 26.50%, 18.93%, 24.22% and

6.81%. Delay evaluation respectively gave3.13%, 21.09%, 22.65%, 21.88%, 10.94%, and 20.31%. These results show that the hybrid architecture achieves lower packet-loss probability and better latency than similar existing schedulers. Comparison of the impact of different coding schemes on the performance of the proposed model respectively gave packet loss evaluation of 60%, 33.33% and 6.67% for G.711, G.726 and G.729. Delay evaluation respectively gave34.19%, 33.86% and 31.94%.These results show that the G.729 coding scheme offers lowerpacket loss and better end-to-end delay at 8kbps bit rate and therefore guarantees better network QoS performance. These facts further validate the optimal performance of the proposed QoS modelwhen high volumes of business/mission-critical data and voice traffic flows are transmitted in constrained-bandwidth VoIP networks. This work has confirmed the fact that only a hybrid scheme can effectively solve the transmission impairment factors of the converged networks. The proposed model was designed in a modular form for easy manipulation. It is structured to be simple and easy to understand. The structure also makes the architecture robust and consistent in its operation.The design ensures that adequate precedence is given to all critical data and voice traffics running on the network. Adequate control mechanisms were incorporated in the structured algorithm for optimal network stability, such that the packet arrival rate is less than the maximum departure rate. The token bucket regulates the traffic via the reserved link while the weighted round robin scheduler regulates the traffic via the shared link. The model also ensures that adequate fairness in resource allocation and utilization is maintained. This implies that the optimized performance of the proposed scheme guarantees a graceful tradeoff between priority (to all critical data and voice traffic flows) and fairness (to all network traffics flows) in a constrained-bandwidth VoIP network. The simulation of the proposed architecture was done on a wired-network topology in this paper. A second stage of implementation on wireless-network topology will increase its robustness to more applications in cyber-physical computer systems and enhance its performance on mobile devices.

References

[1] R. Baheti and H. Gill, “Cyber-physical Systems” in The Impact of Control Technology, T. Samad and A. M. Annaswamy (eds.), 2011. Available at: www.ieeecss.org(Document accessed on 07/12/2018)

[2] Y. Zhang, F. Xie, Y. Dong, G. Yang, and X. Zhou, “High Fidelity Virtualization of Cyber-Physical Systems”, International Journal of Modeling, Simulation and Scientific Computing; Vol. 4, No. 2, 2013, pp 1-26. DOI: 10.1142/S1793962313400059

[3] B. Kehoe, S. Patil, P. Abbeel, and K. Goldberg, “A Survey of Research on Cloud Robotics and Automation”, IEEE Transactions on Automation Science and Engineering, Volume 12, Issue: 2, April 2015, pp 398-409. DOI: 10.1109/TASE.2014.2376492

[4] G. Huston, “Quality of Service - Fact or Fiction”, The Internet Protocol Journal, Volume 3, Number 1, ISSN 1944-1134, 2000, pp 27-34.

[5] J. Ellis, C.Pursell and J. Rahman, Voice, Video and Data Network Convergence: Architecture and Design, From VoIP to Wireless, 1sted., Elsevier Inc., 2003, DOI: https://doi.org/10.1016/B978-0-12-236542-3.X5000-8

[6] M. Ahmed, A. T. Litchfield, S. Ahmed, A. Mahmood and M. F. H. Meazi, “VoIP Performance Analysis over IPv4 and IPv6”, International Journal of Computer Network and Information Security, 2014, 11, 2014, pp 43-48.DOI: 10.5815/ijcns.2014.11.06.

[7] G. R. Ash, Traffic Engineering and QoS Optimization of Integrated Voice and Data Networks, 1st ed., Elsevier Inc., 2007, DOI: https://doi.org/10.1016/B978-0-12-370625-6.X5000-3

[8] C. Tu, “Study on QoS Protection Mechanism of VoIP Systems”, International Symposium on Intelligence Information Processing and Trusted Computing, IEEE Computer Society, 2011, pp 151-153.

[9] J. Dike and C.Ani, (2019), “An Optimized Critical Data and Voice Hybrid Scheduler for Cyber-Physical Computer Systems (CPCS) in Constrained-Bandwidth VoIP Networks”, Proceedings of the IEEE 1st International Conference on Mechatronics, Automation and Cyber-Physical Computer Systems, (MAC-2019), Federal University of Technology, Owerri (FUTO), Nigeria, pp 180-186, March 2019.

[10] A. O. Ariyo and M . C. Olaojoyetan, “The Poor State of Broadband in Nigeria: An Impediment to National Development and Globalisation”, Academic Journal of Interdisciplinary Studies, Vol. 2, No. 12, pp 51-60, 2013. DOI:10.5901/ajis.2013.v2n12p51.

[11] J. N. Dike and D. A. Ogbe, "Challenges of Optimal Deployment of Fiber-Optics Technology in the Nigerian Telecommunications Industry", Proc. of the 2016 International Conference and Exhibition on ICT and Telecommunications (ICEIT), Port Harcourt, Nigeria, pp 74-86, Nov. 2016.

[12] A. D. Ogbe and J. N. Dike, "A Critical Review of Fiber-Optics Technology Development and Deployment in the Nigerian Tellecommunications Industry", Proc. of the 3rd International Conference on Electro-Technology for National Development, IEEE NIGERCON 2017, Owerri, Nigeria, pp 153-162, Nov. 2017. DOI: 10.1109/NIGERCON.2017.8281889

[13] Riverbed Technology, "Riverbed Modeller", Riverbed Technology, San Francisco, USA, 2018. Available at: https://www.riverbed.com/gb/products/steelcentral/steelcentral-riverbed-modeler.html

[14] M. T. Gardner, V. S. Frost and D. W. Petr, "Using Optimization to Achieve Efficient Quality of Service in VoIP Networks", Proc. of the IEEE International Conference on Performance, Computing and Communications, April 2003, pp 475-480.

[15] H. Wu, M. Hsieh and H. Lai, "A Novel Low Latency Packet Scheduling for Broadband Applications, vol. 3768. Berlin Heidelberg: Springer-Verlag, pp 1015–1026, 2005.

[16] A. Shawish, J. Xiaohong and H. Susumu, "Efficient Scheduler for the Growing VoIP Traffic", Proc. IEEE 2007 International Conference on Parallel Processing Workshops (ICPPW), Xian, China, 2007. DOI: 10.1109/ICPPW.2007.38

[17] M. J. Fischer, D. M. B.Masi and J. F.Shortle, "Approximating Low Latency Queuing Buffer Latency", Fourth Advanced International Conference on Telecommunications, IEEE Computer Society, 2008, pp 188-194.

[18] M. E. Culverhouse, B. V. Ghita, P. Reynolds and X. Wang, "Optimizing Quality of Service through the Controlled Aggregation of Traffic", Proc. IEEE 2010 International Conference for Internet Technology and Secured Transactions (ICITST), London, UK, Nov. 2010.

[19] X. Y. Li and L.PingPing, "Research and Improvement of Real-time Queue Scheduling Algorithm", Proc. of the 2010 International Forum on Information Technology and Applications (IFITA), volume 01, IEEE Computer Society, Washington, DC, USA, pp 102-104, 2010.

[20] S. G. Chaudhuri, C. S. Kumar and R. V. RajaKumar, "Validation of a DiffServ based QoS Model Implementation for Real-Time Traffic in a Test Bed". Proc. of the IEEE National Conference on Communications (NCC), Kharagpur, India, 2012. **DOI:** 10.1109/NCC.2012.6176841

[21] P. Senniappan and N. Sivasubramaniam, "Enhanced Core Stateless Fair Queuing with Multiple Queue Priority Scheduler", The International Arab Journal of Information Technology, Vol. 11, No. 2, pp 159-167, 2014.

[22] M. H. Miraz, M. A. Ganie, S. A. Molvi, M. Ali and A. H. Hussein, "Simulation and Analysis of Quality of Service (QoS) Parameters of Voice over IP (VoIP) Traffic through Heterogeneous Networks", International Journal of Advanced Computer Science and Applications, Vol. 8, No. 7, pp 242-248, 2017.

[23] J. N. Dike and C. I. Ani, "Algorithmic Analysis of an Efficient Packet Scheduler for Optimizing the QoS of VoIP Networks", International Journal of Computer Applications (IJCA; 0975 - 8887), Volume 97, Number 4, pp 5-12. 2014, DOI: 10.5120/16993-7104.

[24] J. N. Dike and C. I. Ani, "Design and Simulation of Voice and Critical Data Priority Queue (VCDPQ) Scheduler for Constrained-Bandwidth VoIP Networks", International Journal of Scientific and Engineering Research (IJSER), Volume 9, Issue 9, pp 2050-2056, 2018.

[25] J. N. Dike and C. I. Ani, "Design and Analysis of Voice and Critical Data Priority Queue (VCDPQ) Scheduler for Constrained-Bandwidth VoIP Networks", American Journal of Engineering Research (AJER), Volume 7, Issue 10, 154-167, 2018.

[26] Cisco Systems, "Implementing Quality of Service Policies with DSCP", Document ID: 10103, Cisco Systems Inc, 2008a. Available at: http://www.cisco.com/en/US/tech/tk543/tk757/technologies_tech_note09186a00800949f2.shtml and http://www.cisco.com/image/gif/paws/10103/dscpvalues.pdf(Document accessed on 07/12/2018).

[27] Cisco Systems, "Quality of Service - The Differentiated Services Model", Cisco Systems Inc., 2008b. Available at: http://www.cisco.com/en/US/prod/collateral/iosswrel/ps6537/ps6558/ps6610/product_data_sheet0900aecd8031b36d.html(Document accessed on 07/12/2018).

[28] K. Nisar, A. M. Said and H. Hasbullah, "A Voice Priority Queue (VPQ) Fair Scheduler for the VoIP over WLANs", International Journal on Computer Science and Engineering (IJCSE), Vol. 3, No. 2, pp 506-518, 2011.

[29] A. Demers, S. Keshav and S. Shenker, "Analysis and Simulation of a Fair Queueing Algorithm", ACM SIGCOMM Computer Communication Review, 19(4), pp 1-12, 1989. DOI: 10.1145/75246.75248

[30] S. J. Golestani, "A Self-Clocked Fair Queueing Scheme for Broadband Applications", Proc. of INFOCOM '94 Conference on Computer Communications, Toronto, Ontario, Canada, 1994, pp 636-646. **DOI:** 10.1109/INFCOM.1994.337677

[31] P. Almquist, "Type of service in the internet protocol suite, Request for Comments (Proposed Standard) 1349", Internet Engineering Task Force, 1992.

[32] J. N. Dike and C. I. Ani, "Design and Analysis of Voice and Critical Data Priority Queue (VCDPQ) Scheduler for Constrained-Bandwidth VoIP Networks", American Journal of Engineering Research (AJER), Volume 7, Issue 10, 154-167, Oct 2018.

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