Analysis and Enhancement of QoS in Cognitive Radio Network for Efficient VoIP Performance

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Abstract*—VoIP service demands strong QoS support for enhanced call quality and increased user satisfaction. Deploying VoIP in opportunistic communication models like cognitive radio requires thorough analysis of the factors involved in design and implementation. The objective of this paper is to implement VoIP successfully over a cognitive radio network and maintain the call quality at an acceptable limit. Initially, a model of VoIP is built over cognitive radio network using OPNET Modeler 16.0.A. The QoS parameters of delay, jitter and packet loss are analyzed over a basic cognitive radio cycle in the developed model. Modifications are proposed in the cognitive radio cycle for improvement in call quality and gain in throughput. Extensive simulation studies reflect overall QoS enhancement after application of the modified cycle. Finally, the proposed approach is further enhanced by modifying the VoIP parameters that result in high quality VoIP calls as witnessed from the simulation results.

Keywords-Voice over IP; Quality of Service; Cognitive Radio; OPNET; Performance Enhancement

I. INTRODUCTION

Evolution in the field of wireless communication has witnessed consistently increasing number of users and wider bandwidth requirement of data and multimedia transmitting technologies that have constantly reduced the availability of frequency spectrum. Cognitive radio [1] addresses this problem of spectral congestion by introducing opportunistic usage of the frequency bands that are not heavily occupied by licensed users [1-3]. In cognitive radio, spectrum sensing is done to locate unused spectrum segments and optimally use these segments without harmful interference to the licensed user. Implementation of this technology, therefore, faces unique challenges starting from the capabilities of cognitive radio techniques and the communication protocols that need to be developed for efficient communication to novel spectrum management functionalities such as spectrum sensing, spectrum analysis, spectrum decision as well as spectrum mobility.

Voice over IP (VoIP) [4] has seen considerable success in commercial deployment owing to low maintenance and operational costs and rapid roll out to new services. Considering VoIP as an essential service, supporting more number of voice users while using limited radio resources is a very important issue [5] and is subject to active research that could be a key to the success of future communication systems. Therefore VoIP implementation over cognitive radio network has a bright prospect that must be explored thoroughly. However, real-time constraints imposed by delay sensitive communication paradigm add to the complexities of cognitive radio network design and hence, must be carefully addressed.

Although considerable progress has been made in the research domain of cognitive radio network and its related issues, implementing VoIP over cognitive radio has received less attention. VoIP capacity analysis has been done over a cognitive radio model in [5] through a queuing model based on the MMPP traffic flow and a Markov channel model without any retransmission. This work has been extended over single and multiple channels in [6]. The potential contributions of cognitive radio to spectrum pooling are highlighted in [7] which outline an initial framework for formal radio-etiquette protocols for flexible mobile multimedia communications. Further, adaptive packet scheduling algorithm based on priority based queues for Quality of Service (QoS) maintenance in real-time traffic has been proposed in [8]. However, a comprehensive study of VoIP QoS parameters over a cognitive radio network has not been made till date.

The basic objective of this paper is to build a VoIP model over cognitive radio network, to analyze QoS parameters associated with VoIP call and to enhance call quality through proper modifications. Initially a basic VoIP model is created over cognitive radio network using OPNET Modeler 16.0.A and extensive studies are made to analyze the variation in QoS metrics in Section II. Thereafter, a modification to the existing approach is proposed to enhance quality of VoIP call in Section III. Keeping in mind the scope of cognitive radio network, the proposed approach is further enhanced in Section IV to obtain high quality VoIP calls. Simulation results have been discussed in Sections II, III and IV. Finally the work is summarized in Section V.

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II. Analysis Of Basic Cognitive Radio Cycle

Initially a simple cognitive radio network involving primary and secondary users in a single channel scenario is created in OPNET Modeler 16.0.A [9]. The primary user is modeled to generate traffic at uniform distribution interval between 5 sec to 10 sec. The secondary user is involved in VoIP communication and implements G.711 codec [10]. The primary user is the licensed user who has the priority to use the channel. But it does not always occupy the channel, which leads to the channel being underutilized in the time domain. The secondary user is cognitive radio user who is permitted to use the channel only in the absence of primary user. The cognitive radio cycle is shown in Fig. 1. The secondary user senses the channel during a sensing period and starts its transmission in the transmission period only when primary user is inactive.

A. *Overview of Model*

The node model for secondary user as designed in OPNET Modeler 16.0.A is shown in Fig. 2. VoIP node serves as the application layer node followed by Real-time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP) nodes. The functionalities of each network layer are incorporated in the process model corresponding to each node in the node model. VoIP_sink node acts as sink for packets already received and processed accordingly. The MAC_Controller node acts as link layer node and cooperates with the physical layer node which is involved in sensing, transmission and reception. Spectrum management functionalities such as spectrum sensing and spectrum handoff should work in collaboration with the communication protocols [11] and hence such cross-layer architecture is implemented in this paper.

The MAC_Controller node has the process model as highlighted in Fig. 2. It primarily consists of sense and transmit processes that respectively sense and transmit packets according to the principle stated above. Sensing is performed via single radio architecture [2]. In the single-radio architecture, a specific time slot is allocated for spectrum sensing. Thus only certain accuracy can be guaranteed for spectrum sensing results. Moreover, the spectrum efficiency is decreased as some portion of the available time slot is used for sensing instead of data transmission [12]. The obvious advantage of single radio architecture is its simplicity and lower cost [2] both of which are a must for low cost communication that VoIP promises to offer. Further, sensing is modeled based on radiometry [13] or periodogram which is energy detection based technique. While this may lead to false alarms, the advantage is that it involves low computational and implementation complexities thereby reducing algorithmic delays which may degrade call quality. Moreover it is more generic as receivers do not need any knowledge on primary users' signal [2].

VoIP packets are created as per protocol formats and the functionalities of every field are implemented. Each layer has separate packet format that is modeled in the network by creating RTP, UDP, IP and MAC packet formats. As seen from Fig. 3, the timestamp and sequence number in the RTP packet are used for calculation of latency and packet loss.

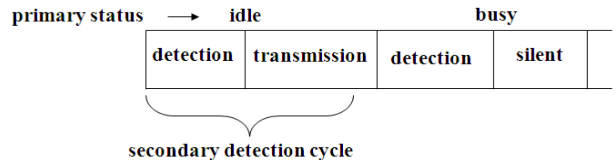

Figure 1. Working Model of Secondary user

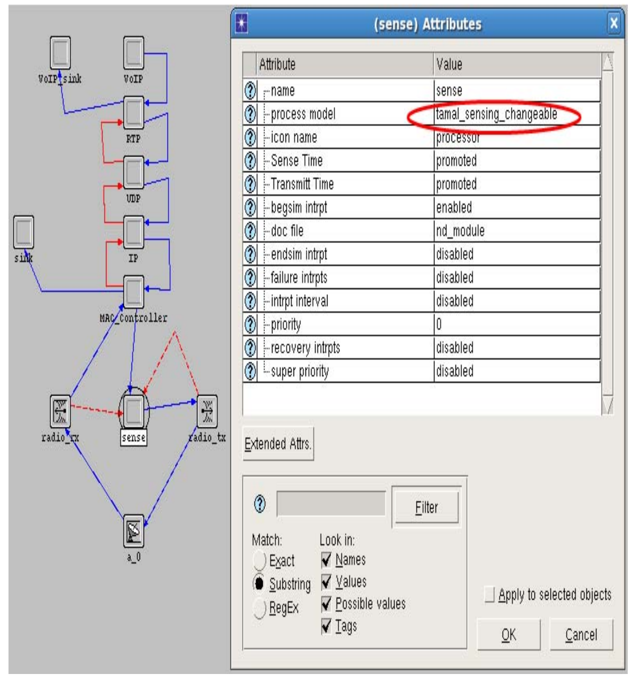

Figure 2. Node Model of VoIP over Cognitive Radio network in OPNET Modeler 16.0.A

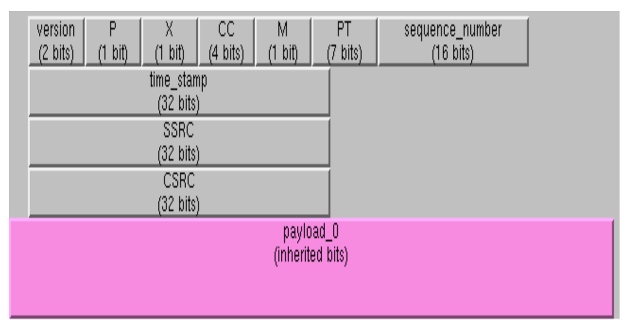

Figure 3. RTP Packet Format in OPNET Modeler 16.0.A

B. *Simulation Results and Discussion*

The primary concern is to ensure that there is no interference or any form of loss with respect to the primary user. Therefore the sensing time and the transmission time for the secondary users are varied accordingly. It is seen from Fig. 4 that the mean end-to-end delay for VoIP calls in the secondary user domain increases with increase in sensing period as more number of packets has to wait for transmission in increased sensing interval. However, increase in secondary transmission period decreases the overall delay which is essential for VoIP calls.

The effect of the cognitive radio cycle is further studied with respect to packet loss in Fig. 5. It is seen that increase in sensing period increases the packet loss of the secondary which further increases with decrease in secondary transmission period.

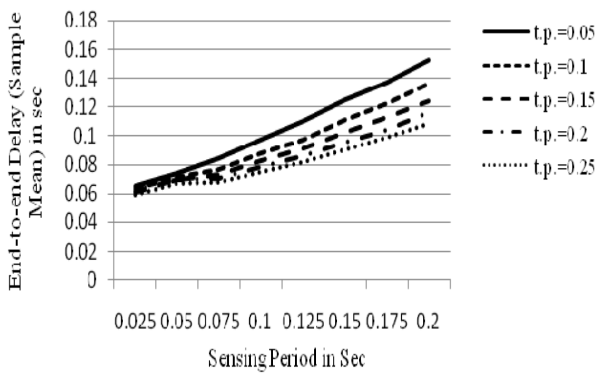

Figure 4. Variation in end-to-end delay (sample mean) with sensing and transmission intervals

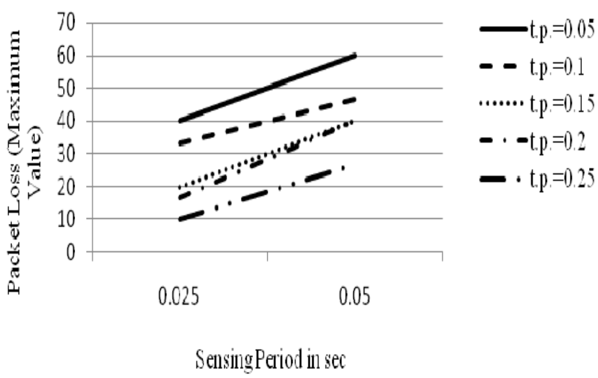

Figure 5. Variation in packet loss (maximum value) with sensing and transmission intervals

Fig. 6 reflects throughput degradation for secondary users with increase in sensing period. Moreover, there is a sharp decline in total traffic received as sensing interval increases. The throughput of secondary users again increases with increase in transmission period as shown in Fig. 7.

Finally, the effect of jitter is analyzed. Jitter is defined as variation in the delay of received packets [14] and controlling it is crucial with respect to VoIP. It is observed from Fig. 8 that jitter increases with increase in sensing period and decrease in secondary transmission period.

Based on the analyzed results, it is clear that sensing and transmission intervals have profound effect on the QoS of VoIP calls. It is evident that the sensing time must not be decreased beyond a certain threshold to avoid interference with the primary user and subsequent loss of information. There are two solutions to this problem. Increasing the sensing period is one solution, which results in low throughput of the secondary users. A short sensing and transmission cycle is the other solution that increases the chances of jitter in voice traffic.

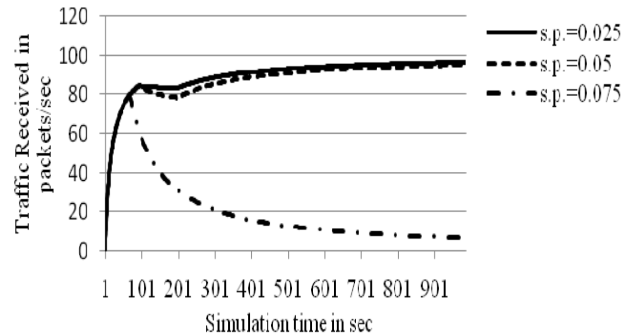

Figure 6. Variation in traffic received with time for varying sensing intervals

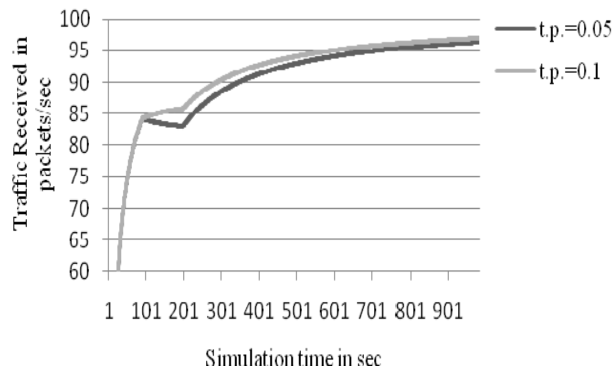

Figure 7. Variation in traffic received with time for varying transmission intervals

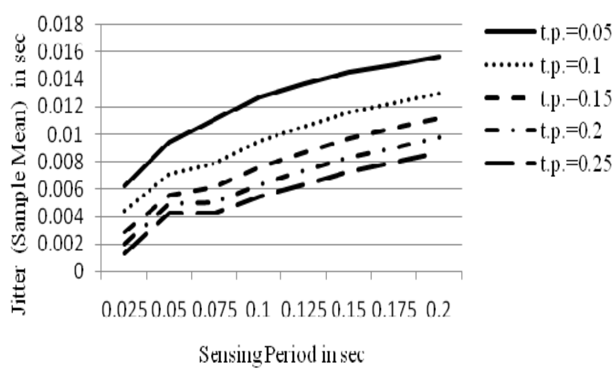

Figure 8. Variation in jitter (sample mean) with sensing and transmission intervals

III. PROPOSED MODIFICATION

In the previous section, it is observed that the sensing and transmission intervals of the secondary users adversely affect the overall QoS of the VoIP calls. The cognitive radio cycle is therefore modified for enhancing the call quality. Instead of having separate sensing and transmission intervals, sensing is performed in the sensing period and also between successive transmissions of packets in the transmission interval.

A. Model Description

The network layer architecture is similar to that of Section II with a minor difference on the interaction between sense node and the transmitter. In the previous scenario, each packet after successful transmission triggers an interrupt from the transmitter to the sense module to send the next packet till the transmission period is over. However, in this scenario, during the secondary transmission period after every packet is sent, the sense module momentarily senses the medium for any primary user before the next transmission. The modified process model for the sensing node is highlighted in Fig. 9 and the principle is described in the flowchart as shown in Fig. 10.

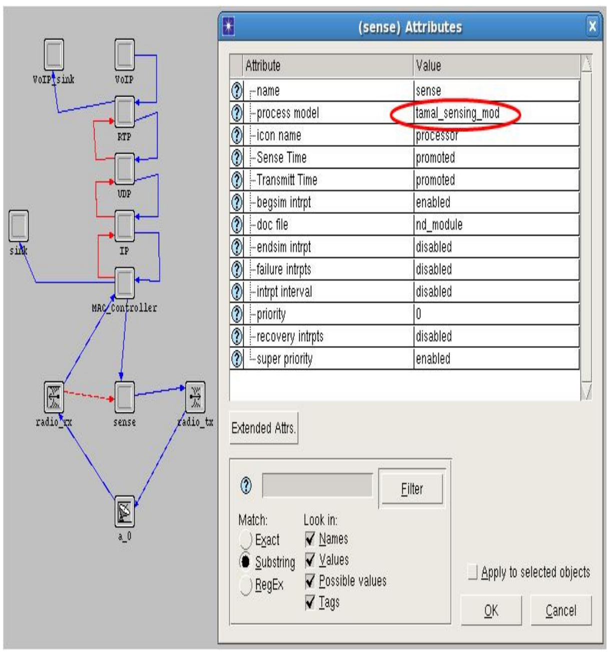

Figure 9 Modified Node Model of VoIP over Cognitive Radio network in OPNET Modeler 16.0.A

B. Analysis of Simulation Results

It is observed from Fig. 11 and Fig. 12 respectively that the end-to-end delay and packet loss are reduced after implementation of the proposed modification. The graphs are plotted with increasing sensing intervals for each transmission period. Further it is observed from Fig. 13 that the standard deviation of the total received traffic from the mean value for secondary users is reduced in this scenario. This results in decrease in jitter as reflected in Fig. 14. Simulation readings illustrate that delay remains within the threshold limit of 150 ms while jitter gradually decreases below 100 ms which is acceptable with respect to voice call.

Therefore, analyzed results point to the fact that the modified cognitive radio model is advantageous especially with respect to VoIP as the overall call quality is enhanced. Implementing the proposed approach reduces the sensing duration while keeping long transmission interval. Such an enhancement ensures maximum throughput of the secondary user without loss of information from the primary user.

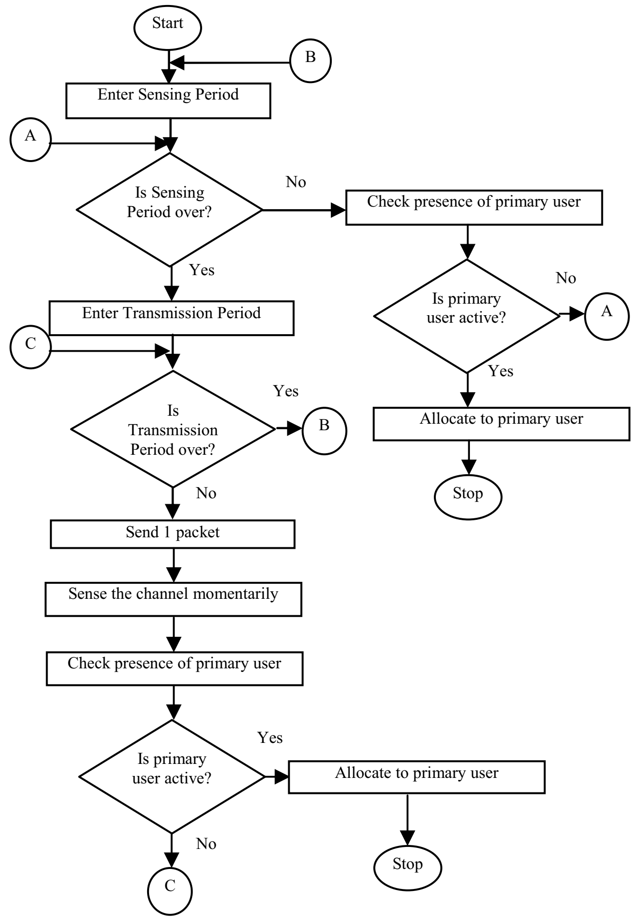

Figure 10. Flowchart depicting the proposed modified approach

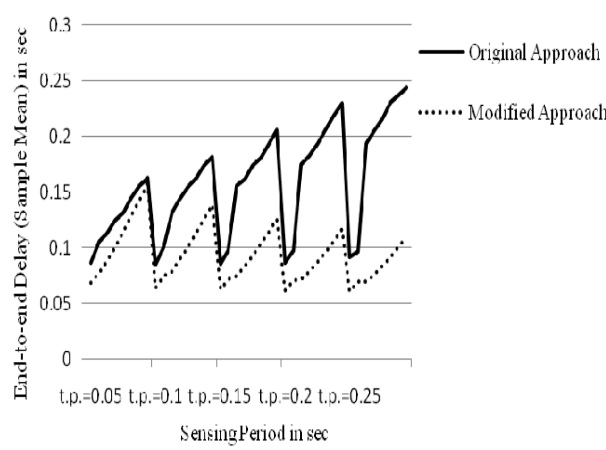

Figure 11. Variation in end-to-end delay (sample mean) with sensing and transmission intervals for the original and the modified approach

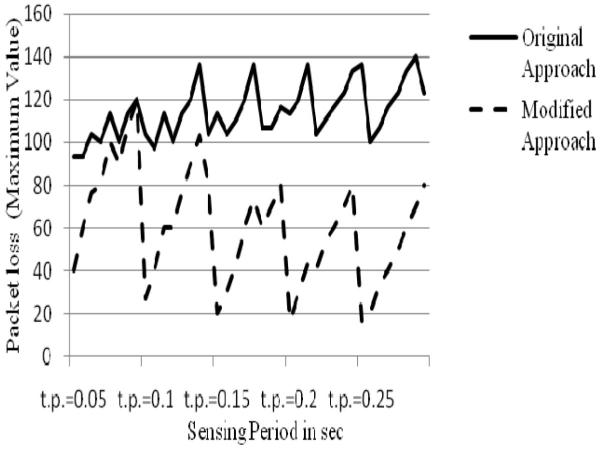

Figure 12. Variation in packet loss (maximum value) with sensing and transmission intervals for the original and the modified approach

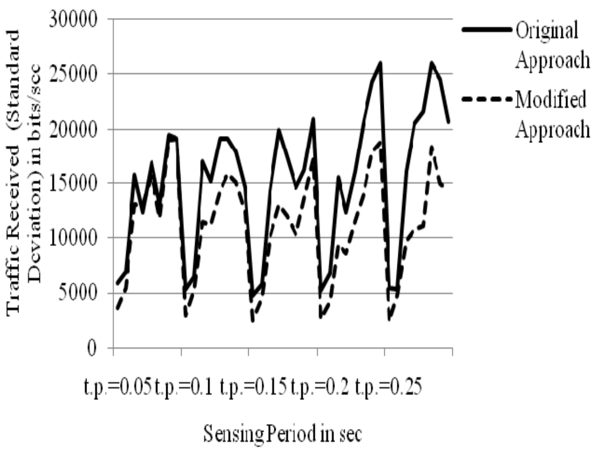

Figure 13. Variation in traffic received (standard deviation) with sensing and transmission intervals for the original and the modified approach

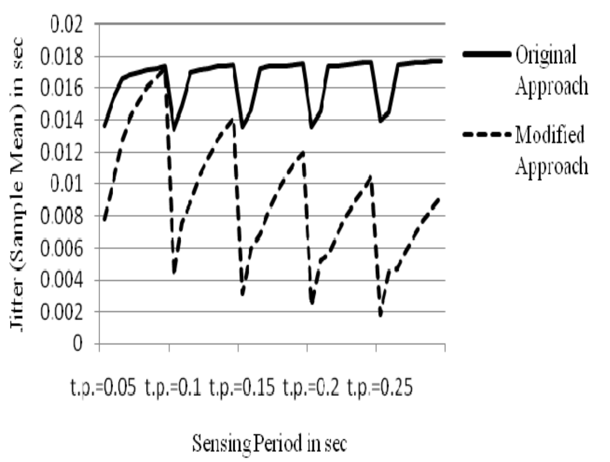

Figure 14. Variation in jitter (sample mean) with sensing and transmission intervals for the original and the modified approach

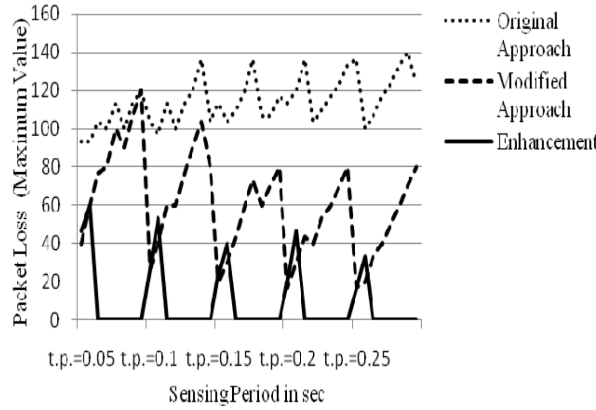

Figure 15. Variation in packet loss (maximum value) with sensing and transmission intervals for the original, modified and enhanced approach

IV. ENHANCEMENT TO THE PROPOSED MODIFICATION

Finally, the proposed approach as discussed in Section III is modified further to enhance the QoS of the VoIP calls from the secondary users. Based on the available primary traffic distribution pattern (for example, as mentioned in [15]), it is possible to send more than a single packet in the transmission period before sensing the channel momentarily for the presence of primary user. If the status of primary users can be predicted to change slowly, sensing frequency (that is, how often cognitive radio should perform spectrum sensing) requirements can be relaxed [2]. To cite an example, the presence of a TV station usually does not change frequently in a geographical area unless a new station starts broadcasting or an existing station goes offline. However, the number of packets from the secondary user to be sent in one transmission slot is totally dependent on the primary user traffic behavior. It implies that a trade-off is required among the involved parameters [16] for performance improvement of VoIP over cognitive radio network.

A. Outline of Model

In our enhancement to the modified approach, multiple packets are sent in one transmission slot before sensing the channel momentarily in the secondary transmission interval. While the node model as described in Section III is kept unaltered, modifications are done in the transmit process of the sense process model. The total number of packets sent at a single transmission slot is increased from 1 to 100 and the simulation is carried out under identical scenarios as done in Section II and Section III.

B. Study of Simulation Results

As observed from Fig. 15, further enhancement to the proposed approach as suggested in Section IV reduces packet loss to a minimum. While the primary user is inactive, more number of packets is sent in one transmission slot before momentary sensing of the channel during the secondary transmission period. Therefore, loss occurs only during low sensing intervals where more number of packets has to contend for the medium. Thus performance enhancement is ensured after application of the proposed modified approach as stated in Section III along with the enhancement stated in this section.

V. Conclusion

In this paper, we have explored the issue of VoIP deployment in cognitive radio systems. VoIP model over a cognitive radio network has been designed in OPNET Modeler 16.0.A by developing highly customized nodes with appropriate coding to include the basic functionalities. Performance analysis reflects deterioration in voice call quality for cognitive network cycle. Modifications are proposed to address the concerns of zero signal interference in case of primary user while achieving maximum throughput with acceptable call quality for the secondary users. Extensive analysis of the proposed enhancements bears a clear testimony to the fact that even with zero tolerance for signal loss in the primary user domain, the quality of service is retained with respect to VoIP for secondary users. While VoIP parameters can be further adapted to such scenarios for performance improvement, the effect of spectrum management and spectrum mobility must also be analyzed for achieving higher success in establishing VoIP over cognitive radio domain.

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References

[1] Federal Communications Commission, “Notice of proposed rule making and order: Facilitating opportunities for flexible, efficient, and reliable spectrum use employing cognitive radio technologies,” ET Docket No. 03-108, Feb. 2005.

[2] T. Y’ucek and H. Arslan, “A survey of spectrum sensing algorithms for cognitive radio applications,” IEEE Communications Surveys and Tutorials, vol.11, no.1, pp.116–130, First Quarter 2009, doi: 10.1109/SURV.2009.090109.

[3] J. Mitola III and G. Q. Maguire, Jr, “Cognitive radio: making software radios more personal,” IEEE Personal Communications, vol. 6, no. 4, pp. 13–18, Aug. 1999, doi: 10.1109/98.788210.

[4] B, Khasnabish, Implementing Voice over IP. Wiley-Interscience, John Wiley & Sons, Inc., 2003.

[5] H. Lee and D. Cho, “VoIP Capacity Analysis in Cognitive Radio System,” IEEE Communication Letters, vol.13, no. 6, pp.393-395, June 2009, doi: 10.1109/LCOMM.2009.082189.

[6] L. Jiang, T. Jiang, Z. Wang and X. He, “VoIP Capacity Analysis in Cognitive Radio System with Single/Multiple Channels,” 6th International Conference on Wireless Communications Networking and Mobile Computing (WiCOM), pp.1-4, 23-25 September 2010, doi: 10.1109/WICOM.2010.5600795.

[7] J. Mitola III, “Cognitive radio for flexible mobile multimedia communication,” in Proc. of IEEE International Workshop on Mobile Multimedia Communications (MoMuC), pp. 3–10, November 1999, doi: 10.1109/MOMUC.1999.819467.

[8] J. Li, B. Xu, Z. Xu, S. Li, and Y. Liu, “Adaptive packet scheduling algorithm for cognitive radio system,” in Proc. ICCT, pp. 1–5, 2006.

[9] O. Inc, “OPNET Modeler.” [Online]. Available: http://www.opnet.com.

[10] ITU-T, G.711: Pulse Code Modulation (PCM) of Voice Frequencies. ITU-T, 1972.

[11] I. F. Akyildiz, W. Y. Lee, M. C. Vuran, and S. Mohanty, “NeXt generation / dynamic spectrum access / cognitive radio wireless networks: A survey,” Computer Networks Journal (Elsevier) 50, pp. 2127- 2159, Sept. 2006.

[12] P. Wang, L. Xiao, S. Zhou, and J. Wang, “Optimization of detection time for channel efficiency in cognitive radio systems,” in Proc. of IEEE Wireless Communication and Networking Conference, pp.111–115, Hong Kong, 11-15 March 2007, doi: 10.1109/WCNC.2007.26.

[13] D. Cabric, S.M. Mishra, and R.W. Brodersen, “Implementation issues in spectrum sensing for cognitive radios,” in Conference Record of the Thirty-Eighth Asilomar Conference on Signals, Systems and Computers, pp. 772- 776, vol.1, 7-10 Nov. 2004, doi: 10.1109/ACSSC.2004.1399240.

[14] “Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms).” [Online]. Available: http://www.cisco.com.

[15] S. D. Jones, E. Jung, X.Liu, N. Merheb, and I-J. Wang, “Characterization of spectrum activities in the U.S. public safety band for opportunistic spectrum access,” in Proc. of 2nd IEEE International Symposium on New Frontiers in Dynamic Spectrum Access Networks, pp. 137–146, Dublin, Ireland, Apr. 2007, doi: 10.1109/DYSPAN.2007.26.

[16] T. Chakraborty, A. Mukhopadhyay, I. S. Misra and S. K. Sanyal, "Optimization technique for configuring IEEE 802.11b access point parameters to improve VoIP performance," In Proc. of 13th International Conference on Computer and Information Technology (ICCIT), pp.561-566, 23-25 Dec. 2010, Dhaka, Bangladesh, doi: 10.1109/ICCITECHN.2010.5723919