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QoS Enhancement Techniques for Efficient VoIP Performance in Cognitive Radio Network

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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 with respect to a single channel. 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. The proposed approach is further enhanced by modifying the VoIP parameters that result in high quality VoIP calls as witnessed from the simulation results. The work is extended in the domain of multiple channels in CRN and an adaptive strategy is introduced in the enhanced version of the proposed algorithm that results in further increase in VoIP call quality. Mathematical analysis of the algorithms provides an outline of optimal selection of related parameters. Finally, a new QoS parameter, *cog_cap* is designed that denotes the cognitive capacity of VoIP calls. Analysis of *cog_cap* for the adaptive algorithm indicates high cognitive capability of VoIP calls thereby proving the efficiency of the proposed technique.**

Keywords: Voice over IP, Quality of Service, Cognitive Radio, OPNET Modeler 16.0.A, Performance Enhancement, Optimization Techniques

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 (CRN) 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.

Deployment of VoIP in CRN network must ensure that network usage is optimized and user satisfaction is maintained with respect to call quality [6]. This clearly requires extensive performance analysis with appropriate tuning of both VoIP and CRN parameters. 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. A comprehensive prototype model of VoIP in CRN has to be developed that will serve as the focal point of analysis and then suitable adaptive policies must be introduced with the aim of satisfying both optimization of resource usage and user satisfaction.

The Quality of Service (QoS) metrics of delay, loss, jitter, Mean Opinion Score (MOS) and R-Factor with respect to VoIP must be thoroughly analyzed for any VoIP performance. However, the method of interaction between the CR features with underlying QoS-supporting mechanisms in applications is not yet completely understood, and system-level evaluation in realistic deployment scenarios is yet to be done [6]. Hence, design of a QoS metric for VoIP in CRN is the need of the hour that will take into consideration both the VoIP parameters and the complexities involved in CRN. This will pave the way for performance analysis of algorithms in VoIP over CRN and lead towards better QoS management.

In this paper, we have developed a comprehensive VoIP model in CRN and achieved enhanced VoIP call quality with significant improvement in spectrum utilization by developing suitable algorithms after carefully analyzing the parameters involved in cognitive radio cycle and VoIP communication. 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 with varying cognitive cycle parameters in a single channel scenario. Thereafter, modifications to the existing approach are proposed to improve the quality of VoIP call in CRN. The work is further extended by developing multiple-channel scenario in the basic model and introducing adaptive strategy in the proposed algorithms to maximize channel utilization with enhanced VoIP call quality. Mathematical analysis of the work further aids in optimal selection of related parameters. Finally, a QoS metric, namely *cog_cap* is designed to monitor VoIP performance in CRN, which measures the cognitive capability of VoIP calls. Application of *cog_cap* in the proposed adaptive algorithm ensures high performance of VoIP users in CRN domain.

II. Motivation

Analysis and optimization of VoIP calls in CRN have witnessed few works in the domain of throughput maximization and efficient spectrum utilization. VoIP capacity analysis, for example, has been done over 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 [7]. The potential contributions of cognitive radio to spectrum pooling are highlighted in [8] 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 QoS maintenance in real-time traffic has been proposed in [9]. However, a comprehensive study of VoIP QoS parameters over a cognitive radio network has not been made till date.

From the limited literature survey of VoIP in CRN, it is hence observed that a complete model of VoIP in CRN is yet to be developed to analyze VoIP performance with respect to several algorithms. While throughput has been the primary performance metric in most works, the effect of delay and jitter has been neglected. Accordingly, the basic motivation behind the work has been focused towards modeling of VoIP in CRN followed by significant improvement of VoIP performance in CRN by adaptive variation in related parameters with support from mathematical analysis and design of QoS metric to quantify the cognitive capability of cognitive users implementing VoIP and applying subsequently the QoS metric to analyze the algorithm. Thus, our work is based on four principal steps namely,

1. Analysis and Optimization of QoS for VoIP in CRN for a single channel scenario.
2. Extension in multiple-channel scenario.
3. Mathematical Formulation.
4. Design and Application of QoS metric for VoIP in CRN.

A model of VoIP in CRN is developed in OPNET Modeler 16.0.A followed by analysis of QoS parameters and subsequent implementation of proposed algorithms to enhance the performance of both VoIP and CRN in Section III. The work has been extended in Section IV in the domain of multiple channels where adaptive strategy is introduced in the proposed algorithm to gain further improvement in VoIP QoS. The entire work is analyzed mathematically in Section V. Section VI features design and application of a QoS metric that denotes the cognitive capability of VoIP users in CRN. Finally, the work is summarized in Section VI.

III. Analysis and Optimization of QoS for VoIP in CRN for a Single Channel Scenario

A. Overview of VoIP Performance in 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 [10]. 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 [11]. 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.

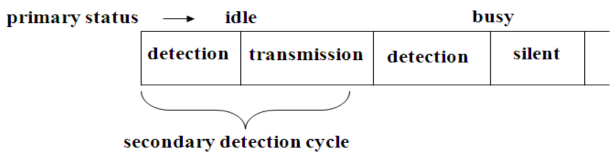

Figure 1. Working Model of Secondary user

1) Description 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 [12] 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 [13]. 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 [14] 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.

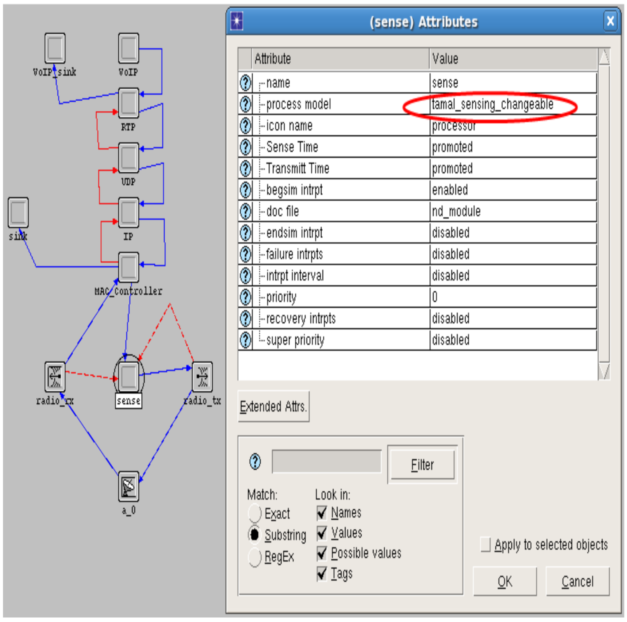

Figure 2. Node Model of VoIP over Cognitive Radio network in OPNET Modeler 16.0.A

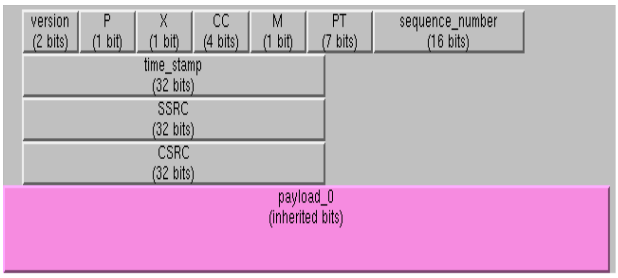

Figure 3. RTP Packet Format in OPNET Modeler 16.0.A

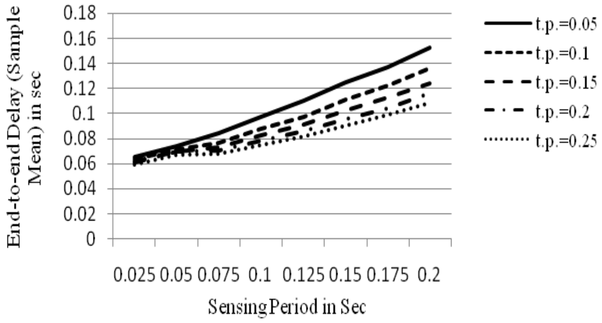

Figure 4. Variation in end-to-end delay (sample mean) with sensing and transmission intervals

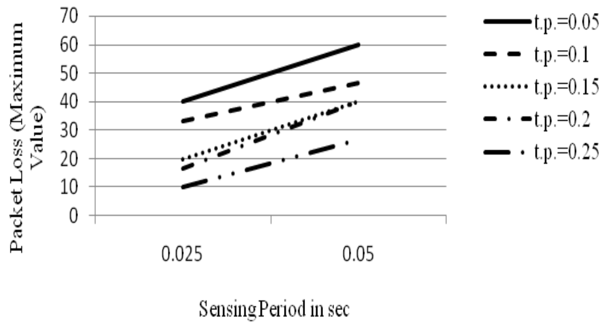

Figure 5. Variation in packet loss (maximum value) with sensing and transmission intervals

2) Simulation Results and Discussion for VoIP performance in the Basic CRN Model

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 user which further increases with decrease in secondary transmission period.

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 [15] 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.

B. *Proposed Modification*

In the previous section III.A, 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 by reducing the sensing interval. Instead of having separate sensing and transmission intervals, sensing is performed in the reduced sensing period and also between successive transmissions of packets in the transmission interval.

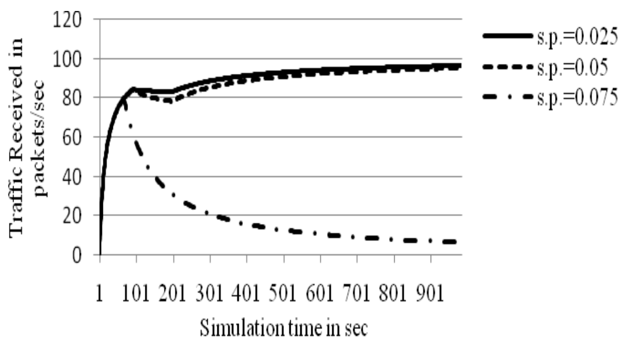

Figure 6. Variation in traffic received with time for varying sensing intervals

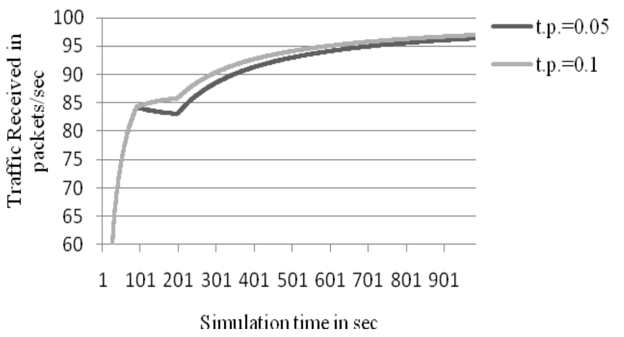

Figure 7. Variation in traffic received with time for varying transmission intervals

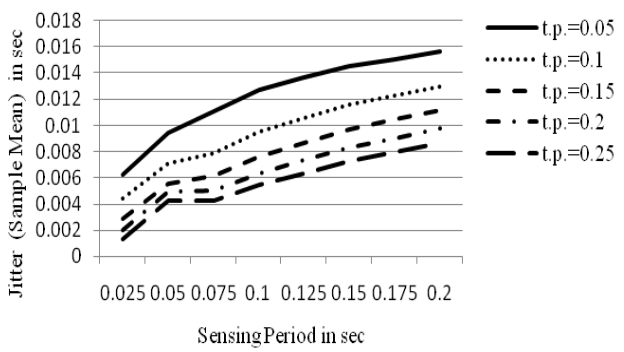

Figure 8. Variation in jitter (sample mean) with sensing and transmission intervals

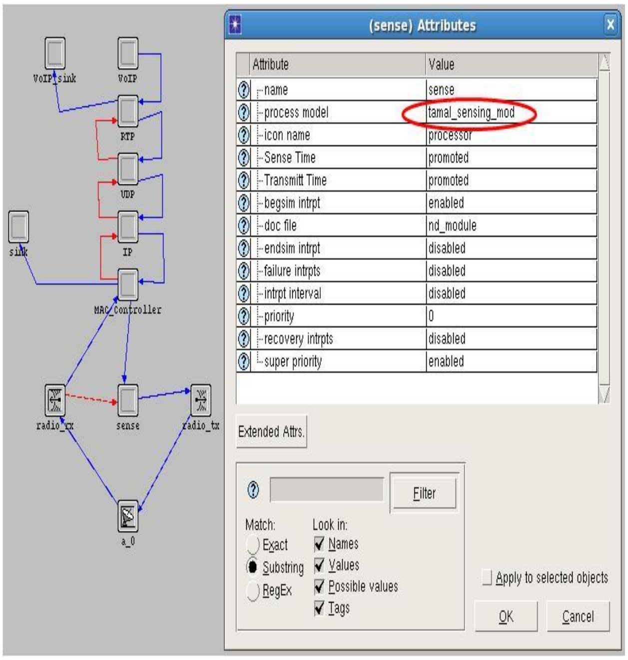

Figure 9. Modified Node Model of VoIP over Cognitive Radio network in OPNET Modeler 16.0.A

1) *Description of the Modified Model*

The network layer architecture is similar to that of Section III.A 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.

2) *Analysis of Simulation Results for Proposed Algorithm*

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.

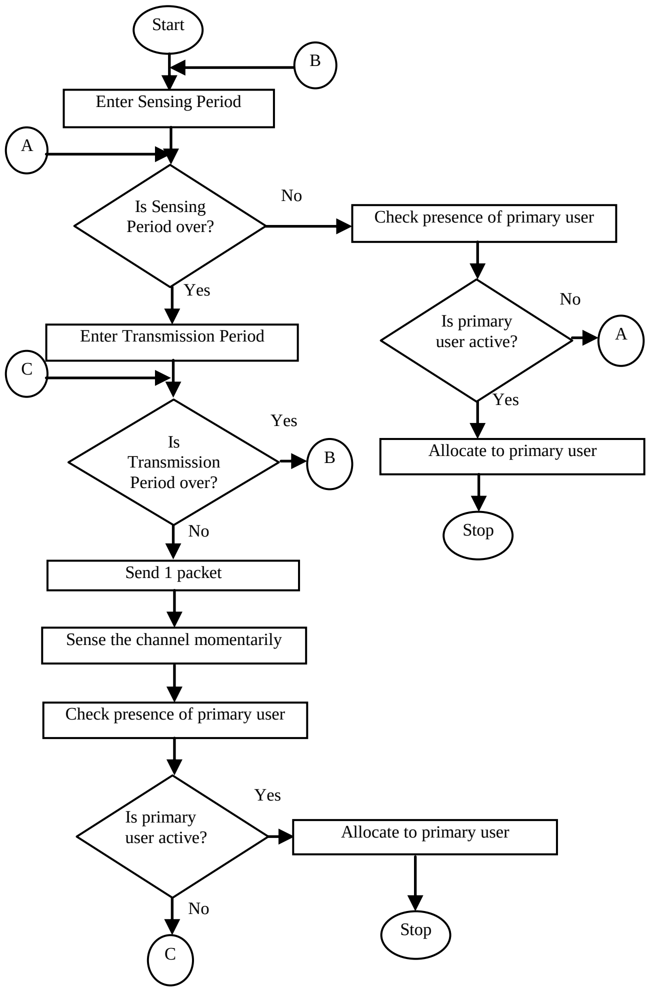

Figure 10. Flowchart depicting proposed modified approach

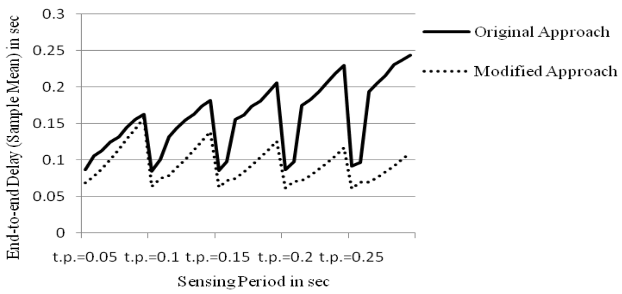

Figure 11. Variation in end-to-end delay (sample mean) with sensing and transmission intervals for the original and the modified approach

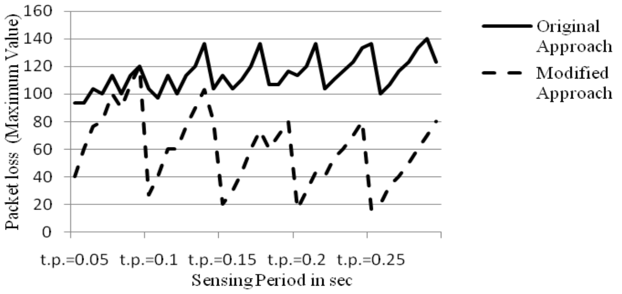

Figure 12. Variation in packet loss (maximum value) with sensing and transmission intervals for the original and the modified approach

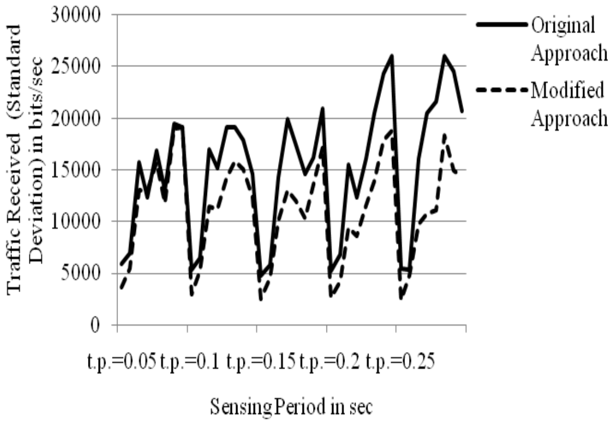

Figure 13. Variation in traffic received (standard deviation) with sensing and transmission intervals for the original and the modified approach

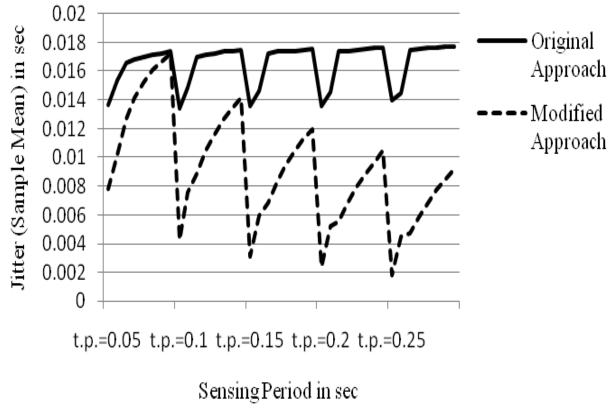

Figure 14. Variation in jitter (sample mean) with sensing and transmission intervals for the original and the modified approach

C. Enhancement To The Proposed Modification

Finally, the proposed approach as discussed in Section III.B 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 [16]), 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 [17] for performance improvement of VoIP over cognitive radio network.

1) Outline of Model for the Enhanced Scenario

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

2) Study of Simulation Results for the Enhanced Algorithm

As observed from Fig. 15, further enhancement to the proposed approach as suggested in Section III.C 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.B along with the enhancement stated in this section.

IV. Analysis and Optimization of QoS for VoIP in CRN for Multiple Channel Scenario

The work is extended in multiple-channel scenario with the assumption that at any particular point of time, atleast one channel is always free for transmission for a given time duration. The issue of allocating a free channel to secondary user is dealt by MAC protocols. As this paper deals with cognitive radio cycle parameters, the assumption is justified. The cognitive radio network developed in OPNET Modeler 16.0.A as described in Section III.B and Section III.C is modified accordingly. The traffic distributions for primary and secondary users as mentioned in Section III are used in this scenario and the algorithm as proposed in Section III.C is implemented in the developed model.

A. Overview of Model

The modified node model for secondary user as designed in OPNET Modeler 16.0.A is shown in Fig. 16. Design of VoIP node along with RTP, UDP and IP nodes and the packet formats is kept similar to the corresponding node models described in Section III. The original node model in Section III.B is modified to include another set of transmitter and receiver that operate in a different channel. This necessitates the design of two sense nodes corresponding to two channels. Therefore, the MAC_Controller node is developed accordingly to switch to the appropriate free channel when the other channel is found to be busy.

The MAC_Controller node has the modified process model. It primarily consists of sense and transmit processes that respectively sense and transmit packets according to the proposed algorithm as stated in Section III.C. Sensing is performed via single radio architecture and is modeled based on the principle of radiometry. Whenever a channel is found to be busy, the other channel is used for transmission and the process goes on accordingly.

B. Methodology of Analysis

As per our objective of creating and developing a comprehensive simulation model for VoIP over CRN, the developed model is used to analyze the performance of VoIP in CRN. The channel throughput is shown in Fig. 17 for each of the two channels from the sender point of view. It is noted that in the presence of primary user, switching to channel 2 takes place and hence transmission goes on simultaneously in both the channels. Both sense and transmit durations are kept at 1 sec. The sudden surge in throughput is observed when the other channel is found to be fully idle even during the momentary sensing periods in the secondary transmission duration as per the proposed algorithm in Section III.C.

Figure 18 shows the time average view of the channel throughput from the receiver point of view. It is worth mentioning that with increase in time, due to increased primary user activity, throughput from channel 1 decreases with rise in throughput in channel2. Finally, the overall throughput of the receiver is shown in Fig. 18 that clearly reflects the fact that there is no considerable dip in throughput at any point of time, thereby indicating the high quality of VoIP transmission.

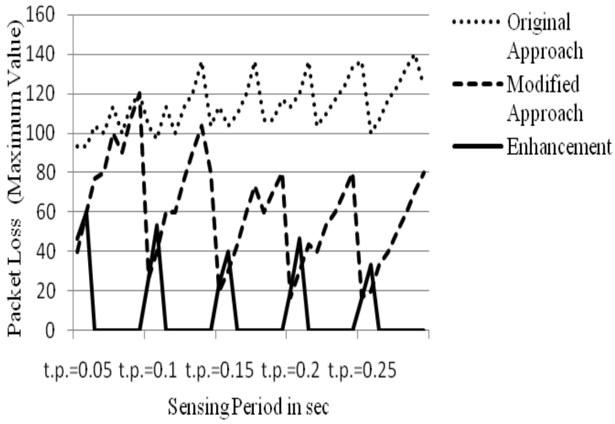

Figure 15. Variation in packet loss (maximum value) with sensing and transmission intervals for the original, modified and enhanced approach

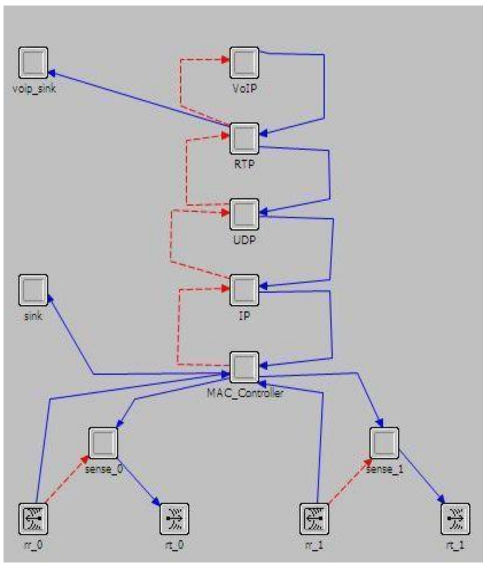

Figure 16. Updated Node Model of VoIP over Cognitive Radio network for multi-channel scenario in OPNET Modeler 16.0.A

In the next step, the channel throughput is analyzed for varying sense and transmission durations in each channel. Fig. 19 and Fig. 20 show the throughput with respect to each channel for varying sense and transmit scenarios respectively. It is observed that the channel throughput increases with increasing transmission duration.

The developed model is used to analyze VoIP performance in CRN following the basic cognitive radio principle as stated in Section IIIA and the proposed algorithm as mentioned in Section IIIC respectively. It is clearly observed in Fig. 21 that overall increase in end-to-end delay in VoIP communication is much less in the enhanced scenario as compared to the basic scenario. There is a delay associated with switching to the other channel whenever the current channel is sensed busy, as this involves active participation and decision making for the MAC_Controller. In the basic CRN, the number of such occurrences with respect to channel switching is more, thereby increasing the delay. As per the proposed algorithm, switching of channels takes place only in the sensing time and hence with reduction in sensing time, the total number of occurrences with respect to channel switching decreases, thereby reducing delay.

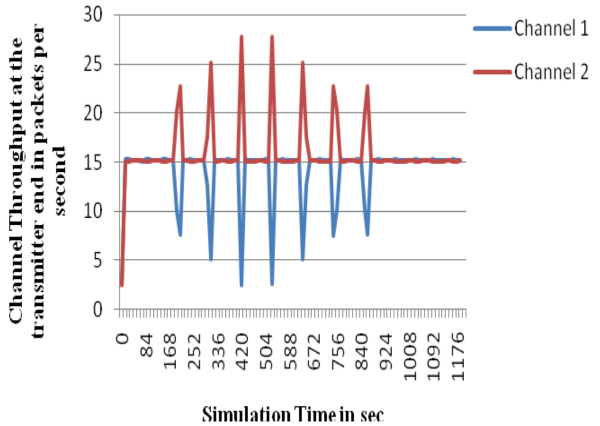

Figure 17. Variation in channel throughput at the transmitter end with respect to 2 channels in CRN for an ongoing VoIP session

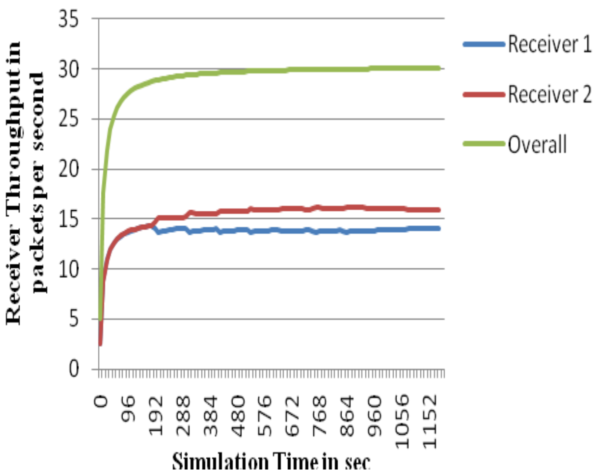

Figure 18. Variation in throughput (time average) at the receiver end with respect to 2 channels in CRN for an ongoing VoIP session

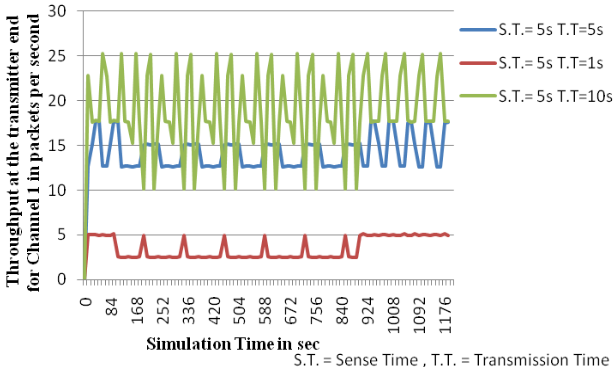

Figure 19. Variation in channel throughput at the transmitter end for channel 1 with varying sensing and transmission intervals in CRN for an ongoing VoIP session

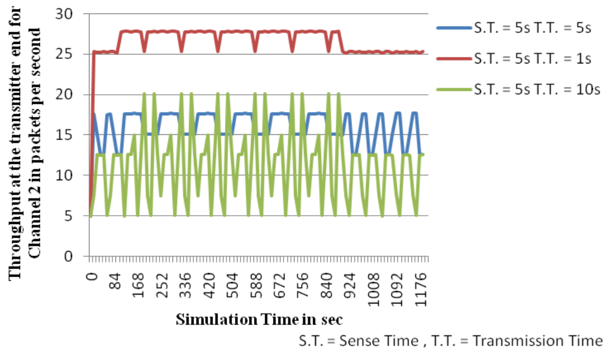

Figure 20. Variation in channel throughput at the transmitter end for channel 2 with varying sensing and transmission intervals in CRN for an ongoing VoIP session

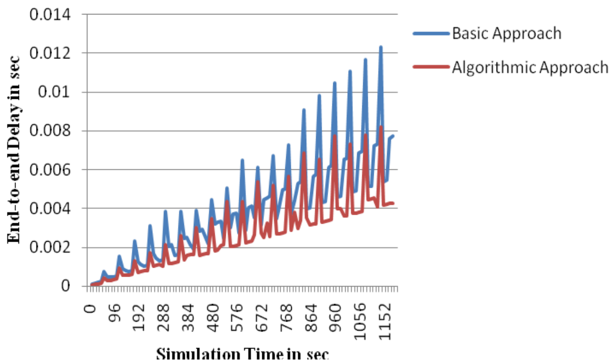

Figure 21. Variation in end-to-end delay for an ongoing VoIP session for the basic and algorithmic approach

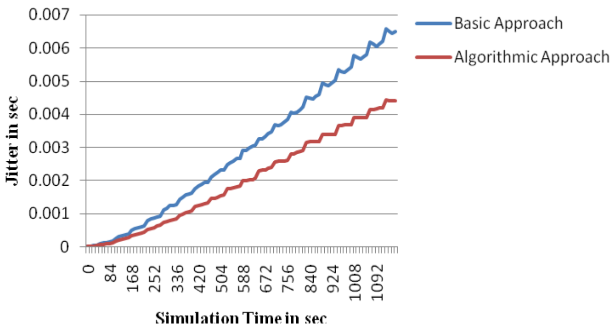

Figure 22. Variation in jitter for an ongoing VoIP session for the basic and algorithmic approach

Moreover, each time a new channel is marked ready for secondary transmission, the delay as described above is introduced, which gets added up as a variable component to the total delay in VoIP transmission. Therefore, in the basic CRN scenario, the total number of variable delay components increases than in the enhanced scenario. As jitter is the difference between successive delays, increase in variable delay components increases jitter and hence the basic CRN witnesses increased jitter condition as compared to the enhanced CRN as observed in Fig.22.

Variation in the rate of received packets provides a clear idea of the packet loss in the model. Fig.23 shows the throughput with respect to secondary user receiver. It is witnessed that the overall throughput decreases in the basic CRN scenario as compared to the enhanced scenario and hence call quality is degraded in the basic CRN.

Thus analysis of delay, jitter, packet loss and throughput bears clear testimony to the fact that implementation of the proposed algorithm results in significant improvement in VoIP performance in CRN.

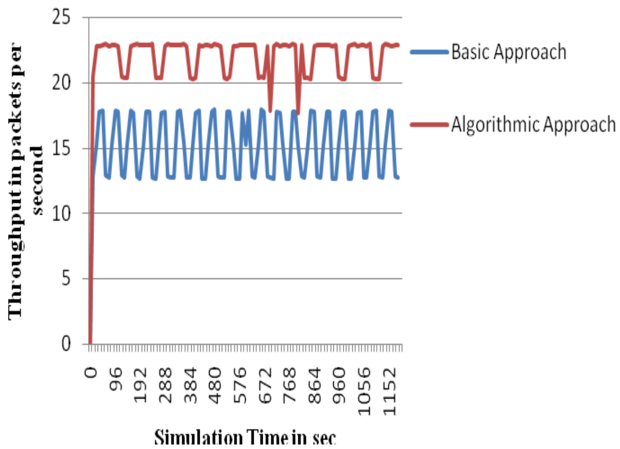

Figure 23. Variation in throughput for an ongoing VoIP session for the basic and algorithmic approach

C. *Proposed Algorithm*

The algorithm along with the enhancement as proposed in Section III aims at reducing the sensing time to a fixed value and increasing the secondary transmission time with momentary sensing slots in the transmission duration. However, having fixed sensing times has its own drawbacks [18]. Firstly, sensing the entire target spectrum continuously may be inefficient and secondly, the channel occupancy state can change rapidly such that the sensing mechanism may fail to keep track of the instantaneous states due to limitations on the sampling time resolution [19].

Therefore, an adaptive strategy is proposed where the sensing and transmission durations are varied according to the channel conditions. A feedback mechanism is deployed that aids in decision making towards runtime configuration of sensing and transmission durations. The feedback process can be implemented with the help of Sender Reports (SR) and Receiver Reports (RR) that are generated during VoIP sessions [20]. The principle of momentary sensing in the transmission duration as proposed in Section III.B is followed for each transmission duration. The adaptive algorithm is described as follows.

1. Start VoIP transmission with a high sensing time (*ts*) and low secondary transmission time (*td*).
2. Set *i*=1.
3. For *i* th cognitive radio cycle,
4. Check for any interference with the primary in td_{i-1} duration. Let it be *Icheck*.
5. If *Icheck*=true, $ts_i = ts_i + K_1$, $td_i = td_i - K_1$.
6. If *Icheck* =false, , $ts_i = ts_i - K_2$, $td_i = td_i + K_2$.
7. If VoIP transmission is not over, $i = i+1$, goto step 8 else goto step 9.
8. Goto step 3.
9. Calculate VoIP QoS and CRN parameters.

D. *Implementation*

The proposed algorithm is hence implemented by suitable modifications to the developed model as described in Section IV. Selection of *ts*, *td*, *K1* and *K2* are crucial for optimum performance of VoIP in CRN for a particular scenario and hence must be chosen carefully. We have chosen two scenarios for low and high primary user presence to analyze VoIP performance for the non-adaptive and adaptive algorithms as proposed in Section III.C and Section IV.C respectively.

1) *Low primary user presence*

In the scenario where the frequency of primary user arrival is less, both the algorithms as devised in Section III.C and Section IV.C respectively are implemented. It is observed from Fig. 24 that the throughput of the channel (where the primary user arrives intermittently) with respect to the secondary VoIP user is higher when the adaptive strategy is implemented. On the other hand, the throughput remains at an average constant value for the non-adaptive strategy even for low primary user activity. The variation of end-to-end delay is depicted in Fig. 25. It is observed that the increase in delay for the adaptive algorithm is less than that of the non-adaptive scenario. The reason is clearly embedded in the fact that adaptive principle minimizes time wasted in channel sensing and hence the time delay. Further, increase in jitter is reduced after implementation of the adaptive algorithm as reflected in Fig. 26.

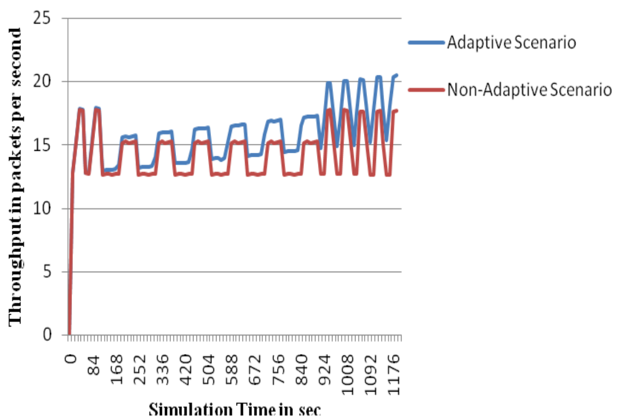

Figure 24. Variation in throughput for an ongoing VoIP session for adaptive and non-adaptive scenarios during low primary user activity

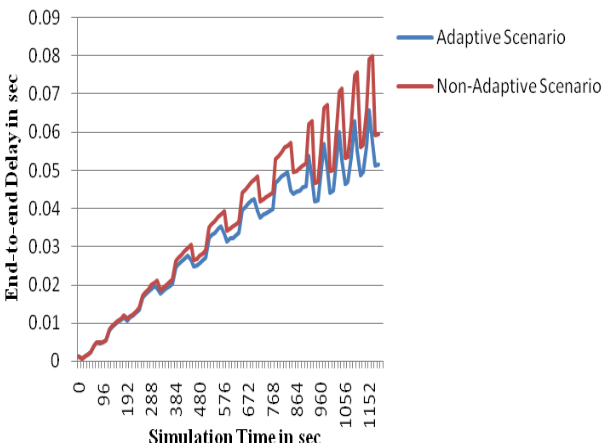

Figure 25. Variation in end-to-end delay for an ongoing VoIP session for adaptive and non-adaptive scenarios during low primary user activity

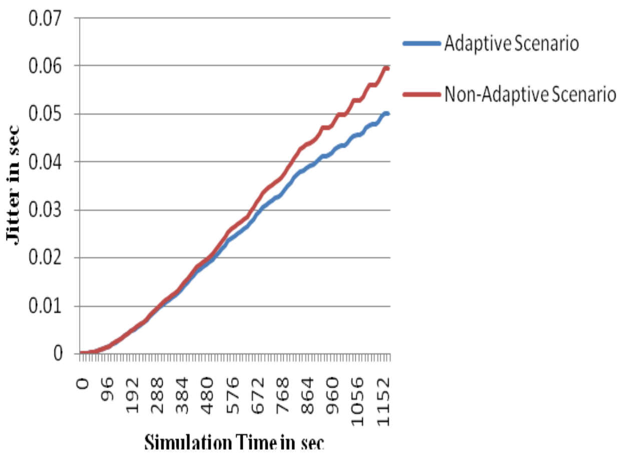

Figure 26. Variation in jitter for an ongoing VoIP session for adaptive and non-adaptive scenarios during low primary user activity

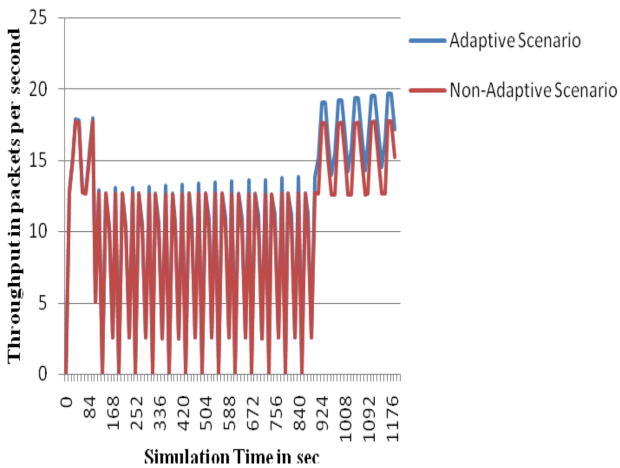

Figure 27. Variation in throughput for an ongoing VoIP session for the adaptive and non-adaptive scenarios during high primary user activity

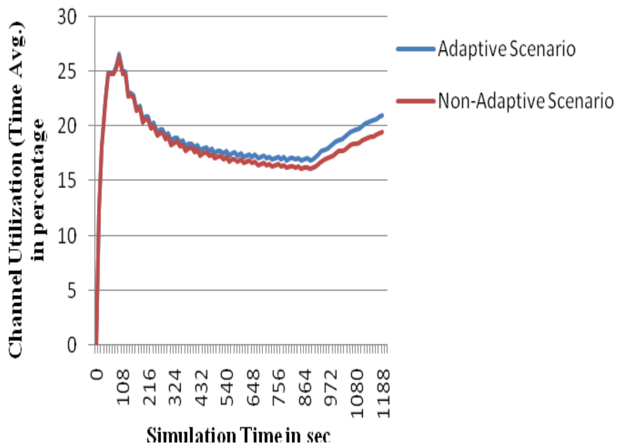

Figure 28. Variation in channel utilization (time avg.) for an ongoing VoIP session for the adaptive and non-adaptive scenarios during high primary user activity

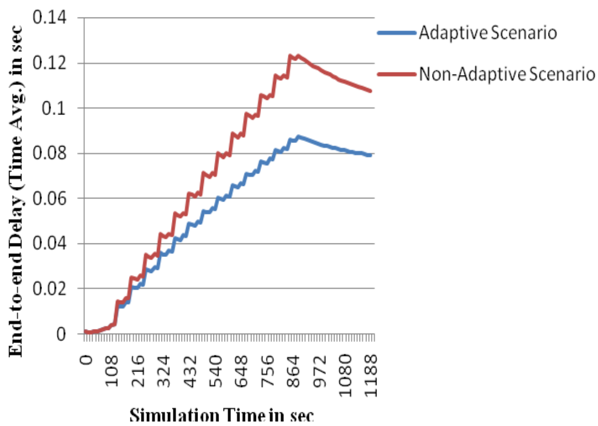

Figure 29. Variation in end-to-end delay (time avg.) for an ongoing VoIP session for the adaptive and non-adaptive scenarios during high primary user activity

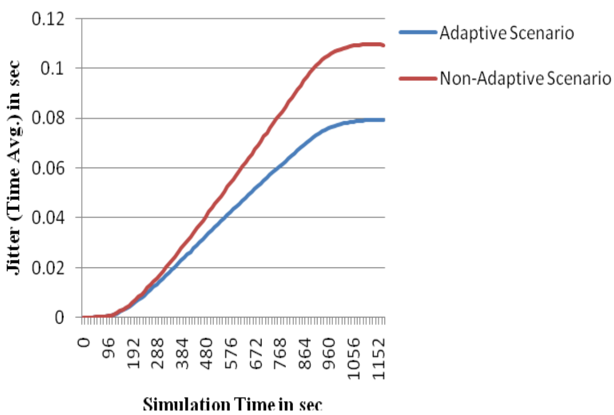

Figure 30. Variation in jitter (time avg.) for an ongoing VoIP session for the adaptive and non-adaptive scenarios during high primary user activity

2) High primary user presence

The algorithms are further analyzed for performance improvement in worst case scenarios having high primary user activity. Figure 27 reflects the throughput of secondary user implementing VoIP calls. It is seen that the adaptive algorithm has a similar performance compared to the non-adaptive algorithm during high primary user activity. Overall, it fares better than the non-adaptive algorithm as far as throughput is concerned. Channel utilization in Fig. 28 depicts that the adaptive algorithm always proves better with respect to channel occupancy compared to the other one. Analysis of delay and jitter from Fig. 29 and Fig. 30 respectively in both the scenarios also indicates efficiency of the adaptive strategy as it helps to contain delay within 100 ms and jitter within 80 ms, clearly reflecting high quality of VoIP call.

Implementation results clearly indicate high quality of VoIP calls after application of the adaptive algorithm, thereby validating its efficiency.

V. Mathematical Formulation

The representation of the basic, modified non-adaptive and adaptive cognitive radio cycle as discussed in Section III.A, IIIB and IVB respectively is shown in Fig. 31.

Initially, the expression of cognitive radio cycle is derived for the adaptive scenario as per the algorithm proposed in Section IVC.

Let t_{cog}, t_{si} and t_{di} be the time equivalent to the total cognitive radio cycle, sensing time and transmission time in the i th cognitive radio cycle respectively.

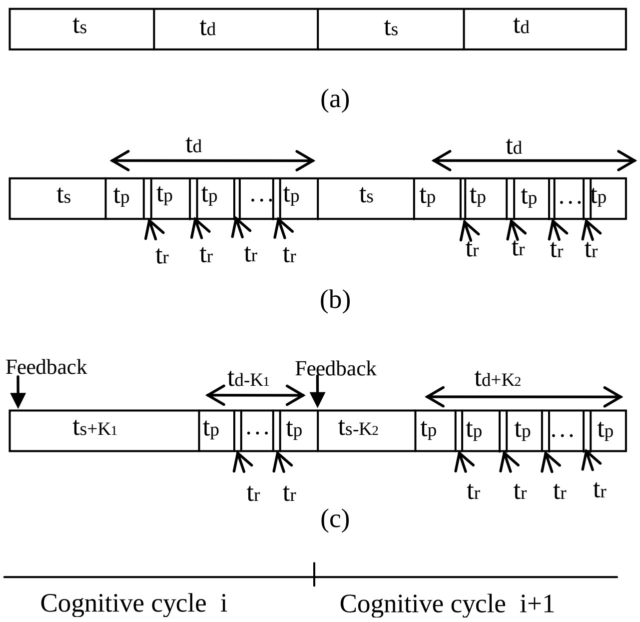

Figure 31. Representation of (a) basic, (b) modified non-adaptive and (c) adaptive cognitive radio cycle

Let P_{psi} denote the probability of primary user detection in the *i* th sensing time i.e. t_{si} and let P_{ctdi} be the probability of detecting collision with the primary user in the *i* th transmission time i.e. t_{di}.

$$\begin{aligned} tcog = & \{t_{s1} + t_{d1} \times (1 - P_{ps1})\} \\ & + \{(t_{s2} + K_1) \times P_{ctd1} + (t_{s2} - K_2) \times (1 - P_{ctd1})\} \\ & + (1 - P_{ps2}) \times \{(t_{d2} - K_1) \times P_{ctd1} + (t_{d2} + K_2) \times (1 - P_{ctd1})\} \\ & \ldots + \{(t_{sn} + K_1) \times P_{ctdn-1} + (t_{sn} - K_2) \times (1 - P_{ctdn-1})\} \\ & + (1 - P_{psn}) \times \{(t_{dn} - K_1) \times P_{ctdn-1} + (t_{dn} + K_2) \times (1 - P_{ctdn-1})\} \end{aligned} \quad (1)$$

Let T_s and T_d be the total sensing and transmission durations for the entire cycle t_{cog} and n be the total number of (T_s + T_d) within t_{cog}.

Deriving expressions for T_s and T_d from (1) above,

$$T_s = t_{s1} + \sum_{i=2}^{n} \{t_{si} + P_{ctdi-1} \times (K_1 + K_2) - K_2\} \quad (2)$$

$$T_d = \sum_{i=1}^{n} \{t_{di} \times (1 - P_{psi})\}$$

$$+ \sum_{i=2}^{n} [(1 - P_{psi}) \times \{K_2 - P_{ctdi-1} \times (K_1 + K_2)\}] \quad (3)$$

As the objective is to maximize t_d, both (2) and (3) are solved further to arrive at the condition expressed in (4).

$$\sum_{i=2}^{n} P_{ctdi-1} \ll \frac{K_2 \times (n-1)}{K_2 + K_1} \quad (4)$$

Thereafter, the expression for the individual transmission period t_d as per the algorithm proposed in Section IIIC is derived while incorporating the effect of the adaptive algorithm as stated in Section IVB.

Let t_p and t_r denote the transmission duration and momentary sensing duration within a particular transmission duration interval t_d of the secondary user. Let N be the total integer value of the number of (t_p + t_r) within t_{di}.

When collision occurs in the *i* th transmission,

$$t_{di} = (N - K_1') \times (t_r + t_p)$$

$$= N_c \times (t_r + t_p) \quad (5)$$

Where $K_1 = K_1' \times (t_p + t_r)$ (6)

When no collision occurs in the *i* th transmission,

$$t_{di} = (N + K_2') \times (t_r + t_p)$$

$$= N_{nc} \times (t_r + t_p) \quad (7)$$

Where $K_2 = K_2' \times (t_p + t_r)$ (8)

Deriving expression for t_d,

$$\begin{aligned} t_d = & [t_{p1} + t_{r1}] + [\{t_{p2} \times (1 - P_{cr1})\} + \{t_{r2} \times (P_{tp2>0})\}] + \ldots \\ & + [\{t_{pj} \times (1 - P_{crj-1})\} + \{t_{rj} \times (P_{tpj>0})\}] \\ & + [\{t_{pN-1} \times (1 - P_{crN-2})\} + \{t_{rN-1} \times (P_{tpN-1>0})\}] \\ & + [(t_{pN} + t_{rN}) \times (1 - P_{crN-1})] \end{aligned}$$

$$= t_{p1} + t_{r1} + \sum_{j=2}^{N-1} t_{pj} \times \left(1 - P_{crj-1}\right) + \sum_{j=2}^{N-1} t_{rj} \times P_{tpj>0}$$

$$+\left[\left(t_{pN} + t_{rN}\right) \times \left(1 - P_{crN-1}\right)\right] \quad (9)$$

As per the algorithm, $t_{pj} = t_p$ for all j.

Therefore, (9) can be simplified accordingly.

$$t_d = t_p \times \left(1 + \sum_{j=2}^{N}\left(1 - P_{crj-1}\right)\right) + t_r \times \left(2 - P_{crN-1} + \sum_{j=2}^{N-1} P_{tpj>0}\right) \quad (10)$$

Therefore, to maximize t_d, the following conditions must be satisfied.

$$\sum_{j=2}^{N} P_{crj-1} << N \quad (11)$$

$$\sum_{j=2}^{N-1}\left(P_{tpj>0}\right) - P_{crN-1} \approx N - 2 \quad (12)$$

For collision in the (i-1) th case, (11) and (12) can be expressed as follows.

$$\sum_{j=2}^{N_c} P_{crj-1} << N_c \quad (13)$$

$$\sum_{j=2}^{N_c-1}\left(P_{tpj>0}\right) - P_{crN_c-1} \approx N_c - 2 \quad (14)$$

For no collision in the (i-1) th case, (11) and (12) can be expressed as follows.

$$\sum_{j=2}^{N_{nc}} P_{crj-1} << N_{nc} \quad (15)$$

$$\sum_{j=2}^{N_{nc}-1}\left(P_{tpj>0}\right) - P_{crN_c-1} \approx N_{nc} - 2 \quad (16)$$

Expanding (4),

$$P_{ctd1} + P_{ctd2} + P_{ctd3} + ...P_{ctdn-1} << \frac{K_2 \times (n-1)}{K_1 + K_2} \quad (17)$$

As collision in t_d for one cycle affects the transmission duration of the next cycle, the total number of momentary sense periods in the transmission duration becomes a function of the probability of collision in the earlier cycle.

Therefore, (17) is modified accordingly.

$$f_1\left(P_{cr1}, P_{cr2}, ...P_{crN-1}\right) + f_2\left(P_{cr1}, P_{cr2}, ...P_{crg2(f1)}\right) +$$

$$f_3\left(P_{cr1}, P_{cr2}, ...P_{crg3(f2)}\right) + ... + f_{n-1}\left(P_{cr1}, P_{cr2}, ...P_{crgn-1(fn-2)}\right)$$

$$<< \frac{K_2 \times (n-1)}{K_1 + K_2}$$

$$\Rightarrow f_1\left(P_{cr1}, P_{cr2}, ...P_{crN-1}\right) + \sum_{l=2}^{n-1} f_l\left(\left(P_{cr1}, P_{cr2}, ...P_{crgl(fl-1)}\right)\right)$$

$$<< \frac{K_2 \times (n-1)}{K_1 + K_2} \quad (18)$$

Where $g_l(f_{l-1})$ is a functional relationship whose value is an integer and is expressed as follows.

$g_l(f_{l-1}) = N_c$ for collision in the previous t_s+t_d cycle. (19)

$g_l(f_{l-1}) = N_{nc}$ in absence of collision. (20)

Therefore, from (18), it is observed that the condition for maximum transmission duration depends on P_{cri} which in turn depends on (13), (14), (15) and (16). It is quite clear from (18) that optimal selection of K_1 and K_2 is very crucial in terms of maintaining VoIP sessions at a level of sufficient quality.

VI. Design and Application of QoS Metric for VoIP in CRN

Quality is defined as "The totality of features and characteristics of a product or a service that bear on its ability to satisfy stated or implied needs" [21]. A distributed system that provides QoS guarantees to general applications must be based on a QoS framework that consists of a QoS specification taxonomy, and a QoS architecture that integrates the different components in the various layers of the system [22]. Therefore, proper QoS requirements must be understood before implementing any mechanism.

VoIP has its own QoS parameters that judge the quality of the call by taking into account the network and user conditions as well as environmental factors. In the field of cognitive radio network, the overall performance of VoIP is also judged by the amount of opportunistic spectrum utilization. However, many current network architectures address QoS from a provider's point of view and analyze network performance, failing to comprehensively address the quality needs of applications [22]. Thus QoS monitoring in VoIP over CRN requires mechanisms to include the overall performance of both CRN and VoIP. The objective is to design a new QoS parameter that takes into consideration the effect of both VoIP and CRN entities.

QoS characteristics can be defined in terms of other characteristics in two ways namely, by specialization of the existing parameters and by derivation and combination of few parameters. [23] As per the objective, we define a derived QoS parameter by taking the combination of both VoIP and CRN parameters to give an indication of the quality of VoIP performance in CRN. As adaptivity is one of the keywords in CRN [24], sensing and transmission time parameters are devised as being adaptive to the changing primary user traffic in this work. Hence in such an adaptive scenario, the variation in voice quality with channel occupancy is mapped onto a parameter *cog_cap* which is a measure of the cognitive capacity of VoIP. It is worth mentioning that as VoIP QoS parameters are unique for a particular VoIP call, *cog_cap* is unique for VoIP session with respect to a particular channel and measures the quality of VoIP in CRN for that particular channel.

A. *Proposed QoS Design Algorithm*

1. Calculate the R-Factor R of the VoIP call after every t moment in T duration to obtain n readings.
2. Calculate the Channel Occupancy Percentage S.
3. Map both R and S on a scale of 1 to 100.
4. Calculate the Correlation Coefficient r with respect to R and S.
5. Determine its strength s.
6. Calculate the critical value r' from the critical r value table for n and statistical significance of 0.05.
7. If $r > r'$ decrease t and repeat from step 1, else goto step 8.
8. Label r as *cog_cap*.
9. Analyze *cog_cap*.

B. *Discussion of the Algorithm*

R-Factor is calculated according to the formula corresponding to the E-model and is given by (21) [25].

$$R = R_0 - Id - Ie \quad (21)$$

Where R_0=93.2, I_d accounts for echo and delay and I_e = equipment impairment factor.

Channel Occupancy Percentage is calculated by the following equation.

$$S = \frac{\text{TotalTime when Secondary occupies the channel}}{\text{Total Time when there is no primary activity}} \times 100 \quad (22)$$

Correlation Coefficient has to be determined as per the nature of the curve with respect to S and R. In case of linear ones, Pierson Correlation Coefficient for two variables X and Y can be used which is calculated according to the following equation [26] for sample size of N.

$$r = \frac{\sum(X \times Y) - \frac{\sum X \times \sum Y}{N}}{\sqrt{\left(\sum(X)^2 - \frac{(\sum X)^2}{N}\right)} \times \sqrt{\left(\sum(Y)^2 - \frac{(\sum Y)^2}{N}\right)}} \quad (23)$$

Strength determines the efficiency of r. The higher the value, the more the parameters are correlated to each other. Significance, on the other hand, indicates the validity of the obtained results and strongly depends on the sample size. If it is 0.05, the results are deemed insignificant and hence the sample size must be increased. Hence the value of r must be greater than the critical value of r corresponding to the significance of 0.05. Strength of r is determined by (24).

$$s = |r|^2 \quad (24)$$

Finally, *cog_cap* gives an idea of the performance of VoIP in CRN. An increasing positive value of *cog_cap* indicates strong correlation between quality of VoIP call and channel occupancy percentage, thereby denoting highly efficient cognitive radio capabilities of VoIP users. Similarly, the more *cog_cap* tends to be 0 and negative thereafter, the more the cognitive capability of VoIP users decreases.

C. *Application of the Designed QoS Metric*

The designed QoS parameter *cog_cap* is hence applied to proposed adaptive algorithm in order to measure the cognitive capability of the VoIP user implementing the adaptive policy. Initially, the R-Factor and the channel utilization percentage at various instants of an ongoing call are measured and after proper scaling mapped onto a graph. As the trend is almost linear as observed from Fig. 32, as per the proposed algorithm, Pearson's correlation coefficient is applied.

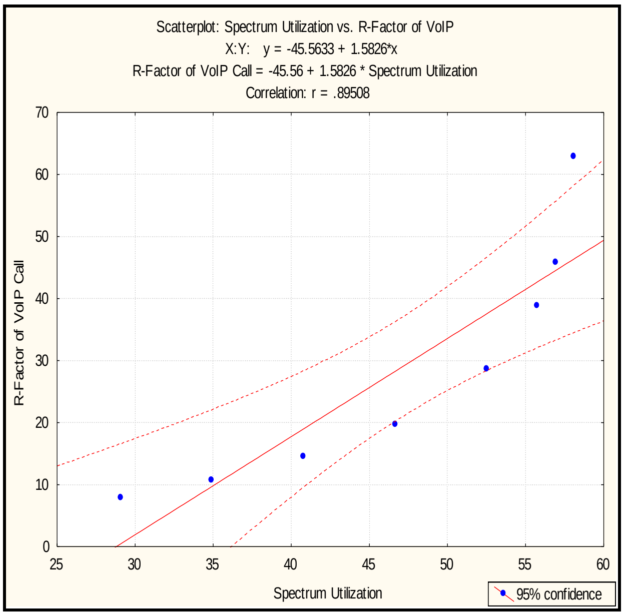

Figure 32. Calculation of *cog_cap* for the proposed adaptive algorithm

It is observed that *cog_cap* is 0.89 having strength 0.79 indicating that as channel utilization increases, there is 79 percent increase in VoIP call quality. The critical value of r corresponding to statistical significance of 0.05 for a sample size of 8 is 0.707 which is less than the obtained value of 0.89. Hence the result is statistically significant. Thus the proposed algorithm is effective in implementing an efficient VoIP system in CRN.

VII. Conclusion

In this paper, we have addressed performance optimization in VoIP over CRN and implemented it in several comprehensive models developed in OPNET Modeler 16.0.A. Firstly, 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 basic 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 in a single channel scenario. 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. Secondly, the model design is extended to include the effect of multiple channels on VoIP performance. Adaptive strategy is introduced in the proposed algorithm which results in further improvement in VoIP call quality as witnessed from the simulation results. Mathematical analysis of the algorithms in the third phase reflects the importance of optimal selection of related parameters towards attainment of the overall objective. Finally, in the fourth phase, a QoS parameter, *cog_cap* is designed that measures the cognitive capacity of the VoIP calls implemented by the secondary users. Application of *cog_cap* to the adaptive algorithm provides highly efficient results, thereby reflecting successful deployment of VoIP in CRN. 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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