Providing QoS to Secondary Users Employing VoIP Applications in Cognitive Radio Networks

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***Abstract*— Quality of Service in Cognitive Radio is an open area for researches. Previous works on this item are classified as either *Quality of Service of Primary User* or *Quality of Service of Secondary User*. The work described in this paper is related to the latter. We worked on a popular application which is Voice over IP. The aim is to provide a sufficient Quality of Service to Secondary Users employing Voice over IP. For that purpose; calls over a real Application Server with different codecs, and different voice packet size were established; voice packets were collected through a network protocol analyzer, which is *wireshark*; packets were analyzed; and an application layer algorithm has been proposed. Then, a Secondary User with a Voice over IP connection employing the proposed algorithm was simulated considering various arrival patterns of Primary User using the network simulator, *ns-2*. The results obtained confirmed the success of the proposed technique. It is shown that, the cognitive radio applications employing the introduced technique achieve an acceptable Quality of Service level for Voice over IP connections. To the best of our knowledge, this is the first work providing Quality of Service to Secondary Users of Cognitive Radio Networks at application layer.**

Keywords-component; Cognitive Radio; QoS; VoIP

I. INTRODUCTION

There is an increasing spectrum demand because of the uptrend in new bandwidth required technologies. However, users could not be always served since the lack of empty spectrum resource in their location. In the meantime, some portions of the spectrum may be underutilized [1]. Since spectrum is assigned statically in current wireless networks, users can not move to another spectrum although there is an available resource on there. Cognitive Radio Networks (CRN) use dynamic spectrum access. Secondary Users (SUs) sense the spectrum, and they are allowed to use a licensed spectrum if they do not affect the Primary Users (PUs) [2-3].

Works related to Quality of Service (QoS) in CRN are very important for the success of CRN. There are two different QoS research area in CRN:

- QoS of Primary User (PU): Aims to show that in CRN, SUs do not affect the QoS of PU.
- QoS of Secondary User (SU): Providing QoS to SUs in CRN.

The main objective of this study is to provide QoS to SUs of CRN. For this purpose, a widely used application, Voice over IP (VoIP), is chosen. Then, a QoS satisfying VoIP application is aimed in order to provide voice communication to SU in CRN. In this work, an application layer solution is proposed.

The rest of the paper is organized as follows: In Section II, the related works are summarized. VoIP Basics are described in Section III. The technique proposed is presented in Section IV. The simulation environment and the simulation results are given in Section V, followed by conclusion and future works in Section VI.

II. RELATED WORK

Recent researches in QoS of SUs in CRN have been focused on power controls, resource management algorithms and Media Access Control (MAC) designs.

In [4] authors identified every user as selfish and each one tries to achieve its target QoS using least power consumption. They had optimized the problem as non-cooperative game, and analyzed Nash Equilibrium (N.E.). In system model, each user announces its interference regulation price and QoS provisioning price. Then, all of users run the proposed multi-channel power allocation algorithm [4].

In [5], authors worked on resource management algorithms. They proposed a hybrid model named C2net which consists of Integrated Services (IntServ) for high priority flows such as voice, video, and Differentiated Services (DiffServ) for other flows. In [6], a scheduling algorithm is proposed for statistical QoS guarantee over CRN. Cooperative relay node and admission control mechanisms are proposed. In [7], authors investigate a CRN which consists of cluster heads (CHs), and regular sensors. Constant Bit Rate (CBR) and Best Effort (BE) traffic is considered and two different priority algorithms are proposed to provide QoS. In [8], authors analyzed the VoIP capacity and proposed a new method for finding the minimum detection and false-alarm probabilities to ensure the QoS requirement of VoIP users in CRN. They modeled the VoIP traffic as Markov-Modulated Poisson Process (MMPP); channel as two state Markov chain.

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In [9] two MAC schemes are proposed for SU accessing the wireless channel. Then, an analytical model is proposed to derive the voice-service capacity for two MAC schemes.

III. VOIP BASICS

VoIP is the growing technology that allows voice conversations to be carried over the Internet Network [10]. VoIP uses Session Initiation Protocol (SIP) [11] or H323 protocol [12] for signaling. Voice conversations have a “sender” and “receiver” roles. Sender and receiver change the roles in different portions of conversation.

A. *Working Principle of VoIP*

Sender creates an analogue signal on the conversation. These analogue signals are digitized by the use of an encoder [13]. At receiver, incoming packets are placed into a playback buffer to overcome problems caused by late received or non-received packets. These packets decoded using identical decoder. The digital bit stream converted back into an analogue signal and send to the receiver.

B. *VoIP Session Initialization via SIP*

Calling and called party get an agreement on which codec is going to be used, and what will be the packet rate, and packet sizes, in each new call. These are agreed at session initialization. Session initialization is done with either SIP or H323 messages. Here, we investigated SIP. There is a Session Description Protocol (SDP**)** [14] portion in SIP messages. It contains all information that needs to be shared with the other side to negotiate. Supported codecs are given in rtpmap attribute of SIP message. This attribute maps the Real-time Transport Protocol (RTP) payload type number defined in "m=" line to an encoding name, clock rate and encoding parameters. RTP payloads are defined by The Internet Assigned Numbers Authority (IANA) and then standardized by The Internet Engineering Task Force (IETF) [15-16]. Voice packet size is carried in *ptime* attribute.

After session is established, calling or called party can change the negotiated codec or *ptime*. This is done by in-session requests with INVITE or UPDATE messages.

C. *VoIP Packet and Codec Relations*

A VoIP packet consists of 20 bytes Internet Protocol (IP) header, 8 bytes User Datagram Protocol (UDP) header, 12 bytes RTP header and a variable size payload according to used codec as illustrated in Fig. 1 [17].

Popular codecs in VoIP have been identified and their packet size, packet interval have been calculated using (1), (2), (3).

$$\text{CodecBitRate} = \text{CodecSampleSize} / \text{CodecSampleInterval} \quad (1)$$

$$\text{PacketsPerSecond}=\text{CodecBitRate}/ (\text{VoicePayloadSizeInBit}) \quad (2)$$

$$\text{TotalPacketSize} = (\text{Layer2Header}) + (\text{IP/UDP/RTP\_header}) + (\text{VoicePayloadSize}) \quad (3)$$

Calculated codec packet size and interval according to *ptime* value is provided in Table I.

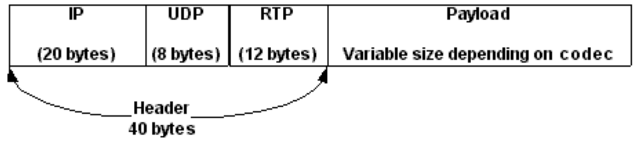

Figure 1. VoIP Packet

D. *QoS Evaluation in VoIP*

IP Networks were built for non-real time applications such as file transfer, email. That is why delay or available bandwidth was not the big concern. Later, IP Networks are started to be used for real time applications and real time applications are relative to delay [18]. In VoIP, packets are created at real time, encoded, packetized, sent over the network, decoded, and listened by other party. Delay, jitter, available bandwidth, as a result QoS, becomes a major concern in real time applications.

The quality of voice is subjective. Users express the quality of a call as “good”, “bad”, “quite good”, or “very bad”. In QoS tests of VoIP, a conversation is listened to users and wanted to quantify the service quality from 1 to 5, 1 being the worst and 5 is the best. The numerical method of expressing voice and video quality is defined as Mean Opinion Score (MOS) [19].

Instead of subjective tests, MOS can be calculated by voice quality effecting factors such as bandwidth, delay, jitter, packet loss, echo or noisy background.

TABLE I. CODEC PROPERTIES

Codec (Defined in rtpmap attribute)	*Voice Payload Size (ms- Defined in ptime attribute)*	*Total packet Size (byte)*
G711	10	126
G711	20	206
G711	30	286
G729	10	56
G729	20	66
G729	30	76
G729	40	86
G729	50	96
G729	60	106
G 723.1	30	70
G 723.1	60	94
G726-32	20	126
G726-32	30	166
G726-32	40	206
G726 -24	20	106
G726 -24	30	136

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The ITU-T E-Model [20] defines an analytic model of voice quality between two connections known as "Voice Transmission Quality from Mouth to Ear" with equipment impairment factor method, and previous transmission rating models. E-model calculates the Rating Factor (R). R value is calculated using the formula below:

$$R = R_O - I_S - I_d - I_{e-eff} + A \qquad (4)$$

where *Ro* is basic signal-to-noise ratio, including noise sources such as circuit noise and room noise, *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal, *Id* is the impairments caused by delay and the effective equipment impairment factor, $I_{e\text{-}eff}$ is the impairments caused by low bit-rate codecs and impairment due to packet-losses of random distribution, *A* is the advantage factor.

MOS value is calculated through R value as below:

$$MOS = \left\{ \begin{array}{ll} R <= 6.5: & 1 \\ 6.5 < R <= 100: & 1 - \frac{7}{100}R + \frac{7}{6250}R^2 - \frac{7}{1000000}R^3 \\ R <= 100: & 4.5 \end{array} \right\} \qquad (5)$$

See Table II for the relation between, R value and MOS.

IV. PROPOSED SOLUTION

After investigation VoIP working principle, and VoIP signaling details, VoIP packets obtained from one of VoIP service provider, Genband Application Server [21] in IP network of Netaş [22] were collected and analyzed. It was noticed that VoIP packet size was constant and these packets were generated periodically. According to SIP Request for Comments (RFC), experimental data shows that packet sizes are relevant to negotiated codec which is given in *rtpmap* attribute in SIP message at session initialization. Packet receive interval is related to *ptime* attribute in SIP message. So, it is obvious that we can model VoIP traffic as CBR traffic.

As stated in Section II, QoS of VoIP is calculated through MOS value. In our proposal, MOS is calculated periodically. The algorithm remembers the previous MOS value. Then, it is compared with the current MOS value.

- If the difference is 0, it means MOS remains the same
- If difference is less than 0, it means MOS decreases
  - If it is -1, then ptime value should be changed.
  - If the change is greater than 1, then codec value is changed
- If difference is greater than 0, it means MOS increases

When difference is 1, ptime value is increased. In this way, packet rate is decreased. Each packet carries more voice packet. But this supplies SU to wait more time for non-received packets.

If difference is more than 1, then codec value is changed to a lower codec. Total packet size is decreased.

TABLE II. RELATION BETWEEN R VALUE AND MOS [28]

R Value (lower limit)	*MOS_{CQE} (lower limit)*	*User Satisfaction*
90	4.34	Very Satisfied
80	4.03	Satisfied
70	3.60	Some users dissatisfied
60	3.10	Many users dissatisfied
50	2.58	Nearly all users dissatisfied

V. RESULTS OBTAINED

The test bed is created as 500X500-grid area in ns-2 [24] which simulates 500 m X 500 m square area. There is 1 SU in simulation and it is simulated with mobile node functionality of ns-2. PU traffic is thought as aggregated traffic, so there is not a number for PU. Existence of PUs is modeled by moving the Secondary User to a place longer than antenna's range. When PU has gone, SU moved back to original place.

In this paper, Cognitive Radio's Spectrum Sensing is out of scope. It is assumed that Spectrum Sensing is done and available channels and slots are specified.

Simulation has been run for 10s. Primary user traffic is modeled as deterministic. The simulation is run under 3 different PU traffic models as listed below:

- ON Period = OFF Period
- ON Period > OFF Period
- ON Period < OFF Period

In our simulations R value is calculated according to delay, jitter and packet loss as in Table III [23].

TABLE III. R VALUE CALCULATION

1	EffectiveLatency = (AverageLatency + Jitter * 2 + 10)
2	# Take the average latency, add jitter, but double the impact to latency then add 10 for protocol latencies
3	**if (EffectiveLatency < 160)**
4	R = 93.2 - (EffectiveLatency / 40)
5	**Else**
6	R = 93.2 - (EffectiveLatency -120/ 10)
7	**Endif**
8	# Deduct 2.5 R values per percentage of packet loss
9	R = R - (PacketLoss * 2.5)

A. ON Period = OFF Period

First the technique is tested when PU ON and OFF periods are equal. ON period and OFF period are taken as 2 s. Fig. 2(a) illustrates PU traffic model. PU arrives to system and transmits data between 2–4 s and 6-8 s.

Fig. 2(b) shows the change in the MOS value of the SU VoIP application when PU traffic is as given in Fig.2(a).

MOS starts to decrease at time 2.0, which is the PU arrival time. Since our algorithm notices a small decrease in MOS, it increases ptime. That's why MOS increases in the next MOS calculation. Similar behavior occurs at time 2.8. When there is a bigger decrease than 1, like at time 7.0, the algorithm changes codec and in the next period MOS increases.

B. *ON Period > OFF Period*

We also tested the traffic model where ON period of PU is bigger than OFF period. PU traffic model is illustrated in Fig. 3(a). PU arrives to system and transmits data between 2–5 s and 7-10 s. This is risky for the SU, since PU remains in the system longer. ON periods are 3 s; while OFF periods are 2 s.

SU MOS changes are given in Fig. 3(b). MOS changes start to decrease after time 2.0, which is the PU arrival time. Since our algorithm notice a small decrease in MOS, it increases ptime. That's why; MOS increased in next MOS calculation period. Same thing occurs in 2.8. When there is a greater decrease, the algorithm changes codec and in next period MOS increased.

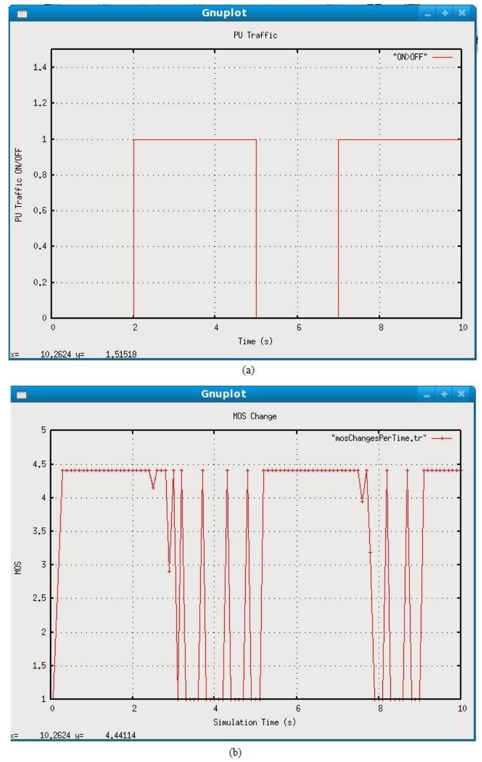

Figure 3. ON Period > OFF Period a) PU traffic b) MOS change on SU

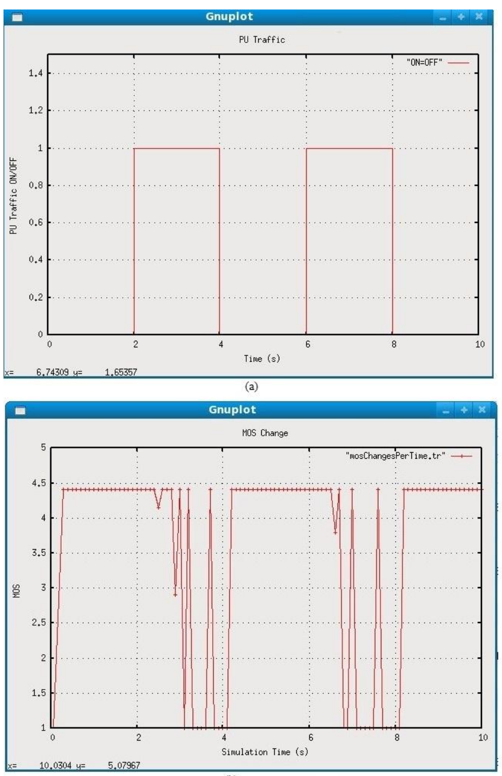

Figure 2. ON Period = OFF Period a) PU traffic b) MOS change on SU

C. *ON Period < OFF Period*

Lately, traffic model of ON period of PU is less than OFF period is tested. Fig. 4(a) illustrates PU traffic model. PU arrives to system and transmits data between 3–5 s and 8-10 s. ON periods are 2 s, while OFF periods are 3 s.

Fig. 4(b) shows the MOS changes in SU according to the behavior of the PU. Changes in MOS start to decrease at time 3.0, which is the PU arrival time. Since our algorithm notice a small decrease in MOS, it increases ptime. That's why; MOS increased in next MOS calculation. Same thing occurs in time 2.8 s. When there is a greater decrease like at time 7.0 s, algorithm changes codec and in the next period MOS increases.

VI. CONCLUSION

This study focuses on the Quality of Service enhancement for the Secondary Users with VoIP connections in Cognitive Radio Networks. One of the most popular applications, Voice over IP is chosen and tried to achieve a satisfying QoS level to the corresponding SUs. For this purpose, first, real VoIP packets are collected from Genband Application Server, and analyzed. After analyzing VoIP packets and VoIP signaling, a solution in application layer is proposed to provide better QoS to SU VoIP applications. Then, we simulate the technique proposed using ns-2. The technique is an application layer approach where the MOS value for the VoIP connection is measured periodically and by chancing the ptime parameter or the codec type according the QoS of the connection is enhanced.

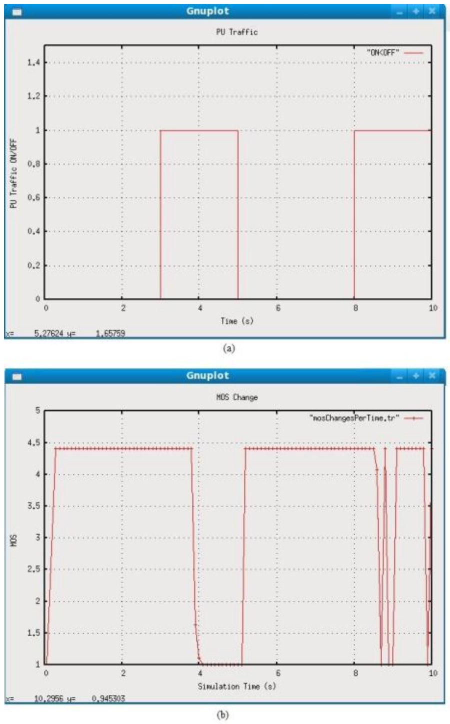

Figure 4. ON Period < OFF Period a) PU traffic b) MOS change on SU.

It is shown that the proposed technique gives promising results for Cognitive Radio Networks by the simulations done in ns-2.

The performance of this algorithm can be further enhanced by not only observing MOS decreases but also observing MOS increases, and changing the codec type or ptime parameter to appropriate values when there is no PU around.

For future work, PUs with different behaviors are going to be employed.

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