ISTANBUL TECHNICAL UNIVERSITY ★ GRADUATE SCHOOL OF SCIENCE ENGINEERING AND TECHNOLOGY

PROVIDING QoS TO SECONDARY USERSEMPLOYING VoIP APPLICATIONSIN COGNITIVE RADIO NETWORKS

M.Sc. THESIS

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Department of ComputerEngineering

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<u>İSTANBUL TEKNİK ÜNİVERSİTESİ ★ FEN BİLİMLERİ ENSTİTÜSÜ</u>

BİLİŞSEL RADYO AĞLARINDA IP ÜZERİNDEN SES İLETEN İKİNCİL KULLANICILARA HİZMET KALİTESİ OLUŞTURULMASI

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OCAK 2012

Esra Hatice Demirtaş, a **M.Sc.** student of ITU **Graduate School of Science and Technology** student ID **504071514** successfully defended the **thesis** entitled "**PROVIDING QoS TO SECONDARY USERS EMPLOYING VoIP APPLICATIONS IN COGNITIVE RADIO NETWORKS**", which she prepared after fulfilling the requirements specified in the associated legislations, before the jury whose signatures are below.

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To my soul mate,

FOREWORD

Firstly, I would like to thank my biggest supporters in my life: my family and my love.

Then, I would like to express my deep appreciation and thanks for my advisor Prof. Sema F. OKTUĞ. This achievement would not have been possible without her support.

Next, I would like to express my graditute to my co-workers and my team leaders in Netaş. I was able to continue my MSc while working, with the support of my managers and my friends. I believe I combined my academic career in MSc and Netaş well together in my thesis.

Finally yet importantly, I thank to TUBITAK for supporting my graduate education with scholarship.

January 2012 Esra Hatice DEMİRTAŞ

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ABBREVIATIONS

AS	**:**Application Server
BS	**:** Base Station
BE	**:** Best Effort
CBR	**:** Constant Bit Rate
CHs	**:** Cluster Heads
CRN	**:** Cognitive Radio Network
DSA	**:** Dynamic Spectrum Access
DiffServ	**:** Differentiated Services
FCC	**:** Federal Communications Commission
InstServ	**:** Integrated Services
ITU	**:** International Telecommunications Union
MAC	**:** Media Access Control
MMPP	**:** Markov-Modulated Poisson Process
MSA	**:** Minislot Assigner
NSIS	**:** Next Steps in Signalling
NSLS	**:** NSIS Signalling Layer Protocol
NTLP	**:** NSIS Transport Layer Protocol
PU	**:** Primary User
QoS	**:** Quality of Service
SI	**:** Scheduling Interval
SDP	**:** Session Description Protocol
SIP	**:** Session Initialization Protocol
SLA	**:** Service Level Agreement
SU	**:** Secondary User
TDM	**:** Time-Division Multiplexing
TDMA	**:** Time-Division Multiple Access
VBR	**:** Variable Bit Rate
VoIP	**:** Voice over IP
2D	**:** Two Dimensional
3D	**:** Three Dimensional

SYMBOLS

D_{enc}	**:** Encoding delay in the system
D_{pack}	**:** Packetization delay
$D_{end\text{-}to\text{-}end}$	**:** End-to-end delay
p_e	**:** Processing delay
f	**:** Frame size
l	**:** The look-ahead delay
CBR	**:** Codec Bit Rate
CSS	**:** Codec sample size
CSI	**:** Codec sample interval
VPS	**:** Voice payload size
PPS	**:** Packets per second
TPS	**:** Total packet size
Ro	**:** Basic signal-to-noise ratio, including noise sources such as circuit noise and room noise
Is	**:** A combination of all impairments which occur more or less simultaneously with the voice signal
Id	**:** The impairments caused by delay and the effective equipment impairment factor
Ie-eff	**:** The impairments caused by low bit-rate codecs and impairment due to packet-losses of random distribution
A	**:** Advantage factor

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PROVIDING QoS TO SECONDARY USERS FOR VOICE OVER IP APPLICATION IN COGNITIVE RADIO

SUMMARY

In this study, providing sufficient quality of service (QoS) in voice over IP (VoIP) application to secondary users (SU) of cognitive radio network (CRN)is aimed. For that purpose, calls over a real application server (Genband) with different codec and different packetization intervals had been established, and voice packets are collected through Ethereal. Then, these collected packets, the Session Initialization Protocol (SIP), which is the protocol for call establishment, and the Session Description Protocol (SDP), which is the protocol for media negotiation,are analyzed. Existing research works about QoS for VoIP in other network types are reviewed and applied to CRN concepts.

After this analysis, an algorithm, which is at application layer, has been proposed to provide an acceptable QoS for VoIP to SU of CRN. In this algorithm, mean opinion score (MOS) that is used for evaluating QoS in VoIP is used. Proposed algorithm is based on measuring MOS value periodically. In each period algorithm compares new MOS value with earlier period MOS value. Algorithm observes decrease and increases in MOS. For decreasing values, algorithm takes action to change either ptime or codec value according to decrease rate in MOS. For increasing values, algorithm puts an indicator that QoS has been increased. It uses this indicator to understand whether same QoS can be maintainedin next 3 periods. If so, algorithm changes ptime and codec value again to use a better codec when there is no primary user (PU) around. With this way, algorithm adapts SU to current network conditions and suply a relevant QoS level.

Then, proposed algorithm is simulated using NS-2 network simulator. MOS change of SU in deterministic and stochastic PU traffic models are tested. Results that we get from simulations confirmed the proposal. If cognitive devices include this algorithm, SUs of CRN can get VoIP in a particular service level. To the best of our knowledge, this is the first research for providing QoS to SUs of CRN at application layer.

BİLİŞSEL RADYO İKİNCİL KULLANICILARINA IP ÜZERİNDEN SES İLETİMİNDE HİZMET KALİTESİ SAĞLANMASI

ÖZET

Yüksekbantgenişliğiihtiyacıduyanteknolojilerinkullanımınınartmasıilebirlikte, frekans ihtiyacı güngeçtikçe artmaktadır.Fakat, kullanıcılar bulundukları konumdaki frekansta boş alan olmadığı için, her frekans ihtiyacı doğduğunda, servis alamayabilirler. Günümüzdeki kablosuz ağlarda, frekanslar statik olarak atandıkları için, başka bir frekansta o anda boş yer olmasına rağmen kullanıcılar boş frekansı kullanamamaktadır. Bu problemin bilişsel radyo ile çözülmesi amaçlanmıştır. Bilişsel radyoda iki tip kullanıcı bulunmaktadır: frekansın sahibi olan lisanslı kullanıcılar ya da başka bir deyişle birincil kullanıcılar ile hattın sahibi olmayan lisanssız ya da diğer bir deyişle ikincil kullanıcılar. Bilişsel Radyo da, ikincil kullanıcıların, birincil kullanıcılar hattı kullanmazken, o frekansı kullanmasına izin verilmektedir.Bu şekilde artan bant genişliği ihtiyacına cevap verebilmek amaçlanmaktadır.

Bilişsel Radyo şu anda araştırma aşamasında olup, herhangi bir kurulmuş ortamı bulunmamaktadır. Her yeni teknolojide olduğu gibi, bilişsel radyolarında, kullanıcılar tarafından kabul görebilmesi için, belirli bir hizmet kalitesini kullanıcılara sağlayabiliyor olması gerekmektedir. Fakat, bilişsel radyo hakkında kaynak taraması yaparken, hizmet kalitesi alanında çok az çalışma olduğu gözlenmiştir. Bu da bizi bilişsel radyolardaki ikincil kullanıcılara hizmet kalitesi alanında çalışmaya motive etmiştir.

İlk olarak Bilişsel Radyo hizmet kalitesi alanında kaynak taraması yapılmıştır. Bu çalışmaları üç ana gruba ayırabiliriz: Güç kontrollü algoritmalar, sistem kaynaklarını ayıran algoritmalar, ya da yeni “ortam erişim kontrolü” (Media Access Control - MAC) geliştirmeleri. Bu çalışmalarda sistemdeki kullanıcıların hepsinin ortak frekans paylaşma algoritmaları kullandığı, aynı MAC protokolünü kullandığı ya da sistemdeki tüm ağ düğümlerinin birbirini bildiği vb. önkoşullar kabul edilmiştir. Bu alanda alt katmanlardan bağımsız, uygulama seviyesinde çalışan, bir araştırma bulunamamıştır. Bu da bizi üst katmanda çalışma konusunda teşvik etmiştir. Bu amaçla yaygın olarak kullanılan ve kullanımı gelecekte daha da artması beklenen IP üzerinden ses iletimi (VoIP) uygulaması seçilmiştir.

VoIP uygulamasının nasıl çalıştığı ve kabul edilmiş standartları incelenmiştir. Daha sonra Netaş ağ ortamında gerçek bir uygulama sunucusu (Genband) üzerinden değişik çözücüler ve değişik paket geliş sıklıkları kullanılarak aramalar yapılmış, bu aramadaki ses paketleri Ethereal programı ile toplanmıştır. Bu ses paketleri ile birlikte, IP üzerinden ses iletiminin ilk aşaması olan, iki kullanıcı arasında oturum kurulmasını sağlayan, oturum başlatma (Session Initialization Protocol – SIP) protokolü ve kullanıcıların hangi tip çözücüleri desteklediklerini birbirlerine iletmelerini sağlayan, medya tanım protokolü (Session Description Protocol – SDP) incelenmiştir.

Bir arama da arayan ve aranan kişi oturum başlatırken o aramada kullanılacak çözücü ve paket geliş sıklığını belirlemektedir. Bu amaçla en çok kullanılan çözücüler ve paket sıklıkları araştırılmış ve bu çözücüler için toplam paket büyüklüğü, paket sıklığı hesaplanmıştır. Yapılan bu hesaplama sonuçları bu çalışma içerisinde sunulmuştur.

Bir sonraki aşama olarak VoIP hizmet kalitesinin nasıl hesaplandığı incelenmiştir. Yapılan araştırmalarda, ortalama görüş puanı (OGP) kavramının kullanıldığı gözlenmiştir. OGP kişisel verilerden oluşan bir puandır. Kullanıcılara yaptıkları bir arama sonrası o aramanın kalitesini 1'den (en kötü) 5'e (en iyi) değerlendirmeleri istenip, tüm kullanıcıların düşünceleriyle OGP hesaplanmaktadır. OGP kişisel verilerle hesaplandığı için bunun servis sağlayıcı tarafında hesaplanabilmesini sağlayan bir formül ihtiyacı doğmuştur. Uluslararası Telekomünikasyon Birliği Telekomünikasyon Standartlaştırma Birimi (ITU-T - International Telecommunications Union Telecommunication Standardization Sector) 2005 yılında bir ağ sisteminin ve bu ağ sisteminde kullanılan ekipmanlara göre bir R değeri hesaplama formülünü tanımlamıştır. Bu R değeri ile OGP değerine nasıl ulaşılacağı da aynı standart içerisinde belirtilmiştir. Genel olarak OGP'yi sistemdeki gecikme, iki paket arasındaki gecikmelerin ağırlıklı ortalaması ve paket kaybı belirlemektedir.

Tüm bu araştırmalar sonrasında, ikincil kullanılara IP üzerinden ses iletimi uygulamasında, yeterli bir servis kalitesini uygulama seviyesinde verebilen bir algoritma geliştirilmesi amaçlanmıştır. Daha önceden kaablolu ve kablosuz ağlarda yapılan VoIP hizmet kalitesi çalışmaları bilişsel radyo uygulamasına uyarlanmıştır.

Önerilen algoritmayı kısaca özetlersek, ikincil kullanıcıların uygulamayı çalıştırdıkları istemcide periyodik olarak OGP puanını etkileyen kavramları hesaplanmaktadır. Bu maksatla istemcilerde periyodik olarak paketlerin gecikmeleri, iki paket arasındaki gecikmelerin ağırlıklı ortalaması ve paket kaybı hesaplanmaktadır. Bu değerlerden her periyotta R değeri hesaplanmakta, R değerinden o periyottaki OGP puanı bulunmaktadır. Sistem bir önceki periyottaki OGP puanını hatırlamakta ve iki periyot arasındaki OGP puan farkına bakarak sistemi gözlemektedir. Eğer iki periyot arası fark sıfırsa, OGP puanının değişmediği;eğer iki periyot arası fark sıfırdan küçükse, OGP puanının düştüğü; eğer iki periyot arası fark sıfırdan büyükse, OGP puanının yükseldiği anlaşılmaktadır. Amacımız OGP puanı düştüğünde bir sonraki periyotta yeniden OGP değerini yükseltmesini sağlayacak bir önlem almaktır. Burada OGP puanını kullanıcıların o aramada kullandıkları çözücü ve paket sıklıkları etkilemektedir. Önceden hesaplamalarını yaptığımız çözücü ve paket sıklıkları algoritmaya doğrudan eklenmiştir. OGP puanındaki düşüş oranına göre paket geliş sıklığı ya da çözücü değiştirerek OGP puanının bir sonraki periyotta yeniden yükselmesi amaçlanmıştır. Eğer OGP puanı arasındaki fark birse, bu durumda algoritma paket sıklığını artırmaktadır. Yani yeni bir paketin kendisine ulaştırılacağı aralığı artırmaktadır. Bu şekilde sistemdeki gecikmelerin OGP puanına etkisi azaltılmaktadır. Eğer OGP puanı arasındaki fark birden de büyükse bu durumda çözücü değiştirmeye gidilmektedir. Daha düşük bir çözücü seçilerek toplam paket boyu azaltılmaktadır.

Önerilen algoritma ns-2 ağ similasyon programında modellenmiştir. Sistemde bir ikincil kullanıcı çifti kablosuz düğüm özelliği ile eklenmiştir. Simülasyonda NS-2 deki 250 m mesafeli anten kullanılmıştır. Birincil kullanıcıların sisteme gelişleri, ikincil kullanıcı çiftini, birbirleriyle konuşamayacakları kadar (kullanılan antenin yayın yapabildiği maksimum mesafenin üzerine) uzağa çekerek modellenmiştir. Birincil kullanıcıların sistemde farklı sürelerde kalışları durumunda ikincil kullanıcılardaki OGP değişimleri test edilmiştir. İlk olarak birincil kullanıcıların sistemde kalış süresinin sistemde olmadığı süreye eşit olduğu model test edilmiştir. Daha sonra birincil kullanıların sistemde kalış süresinin, sistemde olmadığı süreden daha küçük ve daha büyük olduğu modeller test edilmiştir. Daha sonra büyüklük kavramı incelenmiştir. Sistemde kalış süresinin sistemde olmayış süresinin 1.5 katı, 2 katı, 3 katı modelleri test edilmiş ve bu modellerde algoritmadaki bir sorun tespit edilmiştir. Algoritma bu hali ile sadece sistemdeki OGP düşüşlerini incelediği için, her zaman paket sıklığını artırma ya da çözücü düşürme şeklinde önlem almaktadır. Sistemde bu çözücüler sabit sayıda olduğu için, algoritma belli bir noktadan sonra OGP düşüşünü farkettiği halde bir önlem alamamaktadır. Fark edilen bu problem sonucunda algoritmada iyileştirmeye gidilmiştir.

Algoritma artık sadece OGP düşüşlerini değil, OGP yükselişlerini de gözlemlemektedir. Bir OGP yükselişi olduğu zaman, bunu kaydetmekte ve sonraki periyotlarda yükseliş sonrası OGP değerinin korunup korunmadığı gözlemlenmektedir. Eğer sistem bir yükseliş sonrası,devam eden periyotlarda 3 kez üstüste OGP değerini koruyabilirse, sistemdeki sorunun ortadan kalktığı ve daha iyi bir çözücüye geçilebileceği kararlaştırılmaktadır. Sistemde bir düşüş olduğunda sistemin izlediği yol eski algoritma ile aynıdır, sadece yükselişin olmadığı bilgisinin kaydedilmesi eklenmiştir.

Yapılan bu iyileştirme sonrası testler yeniden koşulmuş ve birincil kullanıcıların sistemde kalış süresinin, sistemde olmadığı süreden daha büyük olduğu durumlarda da iyileştirme yapılabildiği gözlenmiştir. Yapılan test sonuçları tez içerinde sunulmuştur.

Daha sonra NS-2'de rastgele trafik modelleri oluşturulmuştur. Bu modellerden birincil kullanıcıların sistemde olup olmadığı hesaplanmış ve algoritma bu trafik modelleriyle de test edilmiştir. Elde edilen test sonuçları tezin ekler bölümünde sunulmuştur.

Son olarak algoritma birincil kullanıcın trafik modeli yığmalı dağılım fonksiyonu şeklindeyken denenmiştir. Yığmalı dağılım fonksiyonundaki ortalama değer parametresi üç farklı şekilde denenmiştir. Bunlar: birincil kullanıcıların sistemde kalma süresinin sistemde olmama süresine eşit olması, birincil kullanıcıların sistemde kalma süresinin sistemde olmama süresinden küçük olması ve birincil kullanıcıların sistemde kalma süresinin sistemde olmama süresinden büyük olmasıdır.

Önceden belirli, rastgeleve yığmalı dağılım fonksiyonu şeklindeki birincil kullanıcıtrafik modelleriyle elde edilen sonuçlar, önerilen algoritmanın doğruluğunu kanıtlamıştır. Bu algoritma ile ikincil kullanıcıların belirli bir servis kalitesinde IP üzerinden ses iletim uygulamasını alabilecekleri gözlenmiştir.

Bu algoritma sadece bilişsel radyo ağlarında değil, aynı zamanda tıkanıklık yaşanan ağlarda, tıkanıklık geçene kadar kullanıcıların hizmet kalitesini etkilememek adına da kullanılabilir. Bu çalışma bilişsel radyo ağlarında uygulama katmanında yapılan ilk çalışma özelliği taşımaktadır.

1. INTRODUCTION

There is an increasing spectrum demand because of the uptrend technologies that require increased bandwidth. However, users may not always be served since the lack of empty spectrum resources in their location. In the meantime, some of the spectrum may be underutilized [1]. Since spectrum is assigned statically in current wireless networks, users cannot move to another spectrum band although there is an available resource on there. This problem is under investigation by dynamic spectrum access in cognitive radio networks (CRN). In CRN, secondary users (SU) are allowed to use a licensed spectrum if there are no primary users (PU) in the channel. SU must observe the channel all the time and evacuate the channel when PU activity is sensed [2-3].

CRN is still under investigation. There is no current deployment yet. Research works on QoS in CRN will drive the initial acceptance and success the CRN. We have noticed that there are few research works on QoS item in CRN. This challenge motivated us to work on QoS item in CRN. According to literature search, there are two different QoS working items in CRN:

1. QoS of PU: These research works investigate that CRN deployment and SU usage will not affect PU existing QoS levels.
2. QoS of SU: These research works investigate providing QoS to SU in CRN.

In this thesis, second research item, providing QoS to SU, is studied.

1.1 Purpose of Thesis

Since QoS is the most important factor for the acceptance of a new technology, this motivated us to work on QoS item in CRN. SU change the spectrum band in time frequently. This brings latency and packet losses, which results as a bad quality. This becomes an important problem on real time services.In this thesis, this problem is studied. The main objective is providing a solution forQoS level to SU in real time applications. For this purpose, a widely used application voice over IP (VoIP) is chosen. We aimed to satisfy QoS level for VoIP application to SU in CRN.

1.2 Contributions of Thesis

This builds on previous and ongoing research works in both CRN and QoS of VoIP.An algorithm is presented to solve the QoS of SU problem in CRN that has been detailed in previous section.This thesis brings a new study framework in QoS of CRN while solving the problem. In contrast to current research works, an application layer solution is proposed. Mean opinion score (MOS) is used to calculate the QoS in VoIP. Proposed algorithm calculates and checks the MOS value periodically. According to the increases or decreased in MOS, algorithm changes the packet time receive interval or data compressing techniques. So, MOS of SUs is maintained in a QoS level with those changes. In addition, this research adapts QoS evaluation of VoIP studies in other networks.

1.3 Organization of the Thesis

The structure of the thesis is as follows: In Section 2, the basics of Cognitive Radio, the literature search results on QoS work item in CRN, are given. Following with Section 3, information about VoIP basics, session description, session initialization, and codecs provided. In addition, QoS evaluation in VoIP explained in same section. In Section 4, an algorithm for QoS in SU is proposed and tested in NS-2. Simulations results provided in same section. Finally, in Section 5, conclusion and future work are stated.

2. COGNITIVE RADIO

2.1 Cognitive Radio Basics

Wirelessmultimedia applications require significant bandwidth. Federal Communications Commission (FCC) has assigned current wireless spectrums on a long-term basis for large geographical regions to service providers [2]. These are called licensed or primary users (PU). However, according to FCC reports, usage in spectrums changes according to time and geographical location as illustrated in Figure 2.1 [1].

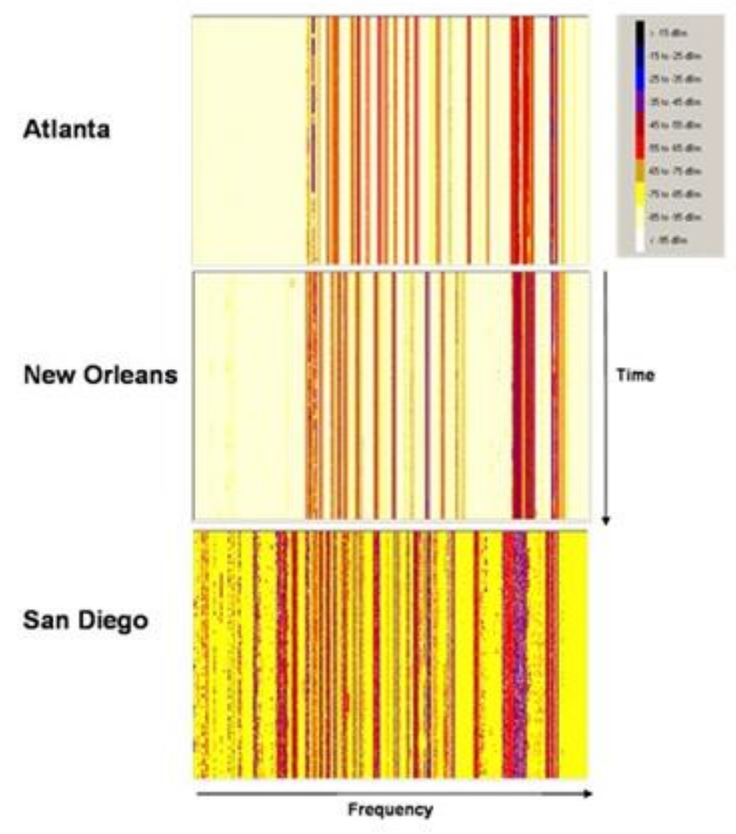


Figure 2.1 : Spectrum usage according to time and location –Adopted from [1]

Spectrum is limited but there is not a smooth distribution in spectrum usage. So, to overcome the inefficiency, cognitive radio networks (CRN) or dynamic spectrum access (DSA) networks have been proposed. In CRN, there are second type users who have no spectrum licensed, also known as secondary users (SU). These users are allowed to use the temporarily unused licensed spectrum, also called spectrum hole or white space, when they do notaffect PU usage [4]. SU have cognitive capability and reconfigurability. Cognitive capability stands for the ability to sense the environment, learn the characteristics of environment, and decide to use. Reconfigurability stands for adapting operating parameters, such as transmission power, frequency, modulation type, communication technology etc., to the variations of the surrounding radio environment [4].

The cognitive radio cycle consists of detecting spectrum holes, selecting the best frequency bands according to used application, coordinating spectrum access with other users and vacating the frequency band when PU appear [2].Figure 2.2illustrates this cognitive radio (CR) cycle with cognitive radio terms.

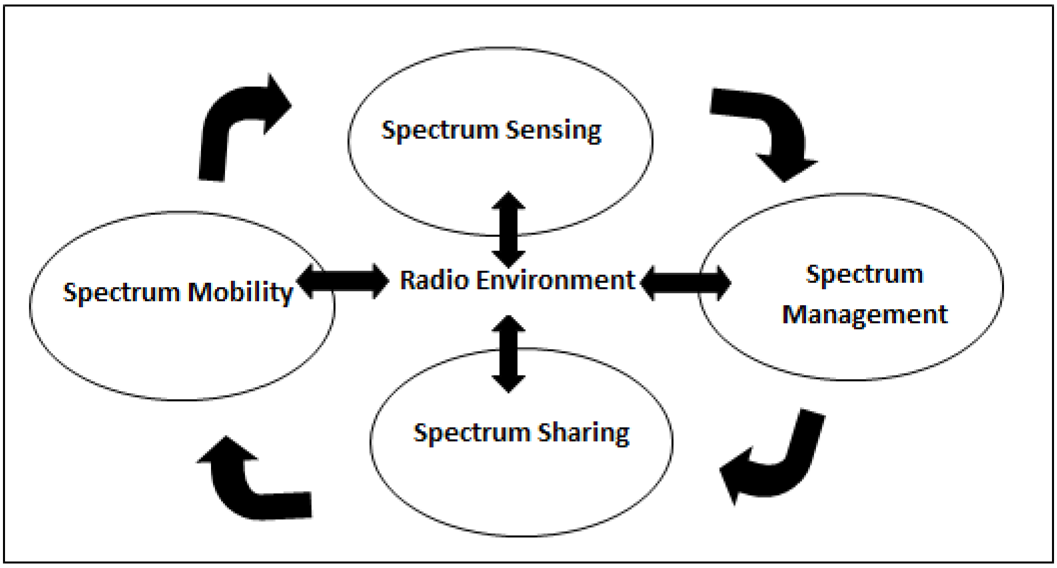


Figure 2.2 : Cognitive radio cycle- Adaptedfrom [2]

As seen fromFigure 2.2main working items in CRcan be grouped into four categories: Spectrum Sensing, Spectrum Management, Spectrum Sharing, and Spectrum Mobility.

2.1.1 Spectrum sensing

Spectrum sensing is used for making necessary observations about its surrounding radio environment, such as the white holes and the presence of PU. Spectrum sensing can be performed in the time, frequency, and spatial domains [2]. Current spectrum sensing algorithms can be summarized as energy, feature, matched filter, interference based, or cooperative detection; learning/reason based sensing, statistical covariance-based sensing [5].

2.1.2 Spectrum management

Spectrum Management is used forselecting the best available channel for required transmission. This work depends on spectrum sensing. All available spectrums sensed by sensing unit are analysed, and channel capacities are calculated. Then according to user QoS requirements, application type etc. one of the available channel is decided for transmission [3].

2.1.3 Spectrum sharing

Spectrum Sharing is used for coordinating access to chosen channel in earlier steps. Because, there can be more than one CR users in the environment and same channel can be chosen for transmission by different CR users. There have been different spectrum sharing algorithms worked: centralised or distributed sharing algorithms, cooperative or non-cooperative algorithms [3].

2.1.4 Spectrum mobility

Spectrum Mobilityis used for changing the working spectrum at SU. This channel change may occur because of current channel conditions becoming worse, or a better channel found for user requirements or the PU appearance in the current channel. In any of these conditions, SU stop transmitting on current channel and move to another available channel. [2-3]

FCC has considered CRN as a promising technology.However, QoS in CRN is still an open area. It is a very challenging work item due to various factors such as dynamical changing of topology, capacity limitations, link variability and multi-hop communications, etc. So, it is becoming crucial to ensure QoS to SU. Based on this open area in CRN, in this thesis, we have focused on employing QoS to SU.

2.2 Literature Review on QoS for VoIP

With the general growth of bandwidth need, QoS to SU in CRN is becoming a popular research area. However, there are less research works in this area currently.

According to literature search, recent research works on this areacan be grouped into three categories: power controls, resource management algorithms and new MAC designs.

In [6], a spectrum sharing algorithm is proposed for distributed multi-channel power allocation problem. The problem was formulated as a non-cooperative game in which each user aims to achieve QoS level using the least power consumption. In the 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. Since each SU uses this spectrum sharing algorithm, authors brought also a QoS guarantee to SUs. However, QoS becomes bounded to spectrum sharing algorithm in this case. In real world, different spectrum sharing algorithms can be run on different SU.

Several authors [7-11] worked on resource management algorithms to supply QoS to SU. They mainly separate resources of system according to QoS levels. WhenSU wants to use the channel, they decide to allow or reject according to QoS level of SU and current system resources. In [7], authors proposed a hybrid model named C^2net which has Integrated Services (InstServ) for high priority flows such as voice, video and Differentiated Services (DiffServ) for other flows. In [8], authors provide a statistical QoS to SUs with relay node concept. High priority packets of SUare sent over another node. Algorithm achieves good results however; all nodes must know the network scheme and this is not realistic for usage in real life. In CRN, SU and PU change dynamically over time.In [9], authors bring sensor network and CRN concepts together. They assign CRN properties to main sensors of sensor network and propose two different algorithms according to packet types. They decide to send which data according to system resources and used algorithm. In this way, they supply a QoS level. In [10], Portfolio Selection Theory, which is originally used in finance, is applied to CRN for QoS problem. In this theory, the optimal wealth allocation is found while minimisingthe portfolio risk for investors. In CRN alignment of this theory, optimal channel allocation to maximisethe throughput is investigated. This will lead to guarantee a statistical QoS. In [11], authors analysed 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 modelled the VoIP traffic as Markov-Modulated Poisson Process (MMPP);

channel as two state Markov chain. With their algorithm, they implement QoS of SU by Base Station (BS). BS decides to send data or not according to user's QoS level and current resource usage.

In [12] a new study areais introduced to solve the problem. Authors design two new MAC schemes;as a result, they supply QoS to SU using this MAC scheme. However, this design requires implementation at all SU clients.

Motivated by these research works and considerations, we noticed that current QoS research worksrely on all SU using same spectrum sharing algorithmor same MAC protocolorneedsknowledge in current network paths, etc. We identified that supplying QoS to SU without these constraints can still be investigated. There isstill opportunities for algorithms that can be easily implemented in any SU at client side.With this motivation, we aimed to supply a QoS level to SU in a specific application. We choose VoIP.Because VoIP will be an essential service in the future since users can more cheaply utilisevoice services. Therefore, supporting a QoS level for VoIP in CRN is an important research area.With our study, we started a new research methodology in QoS of SU in CRN. For this purpose, we also investigated research works that study QoS of VoIP in other network types.

Firstly, we have investigated VoIP models. ITU–T defines the human voice conversations as an ON and OFF model [13]. It is accepted that signals are created in Talkspurt duration (T_t), and not created in Silent duration (T_s). The speech state transition diagram consists of Single Talk, Double Talk, and Mutual Silence. In [14] Authors add the long silence and long burst states to ITU- T definition. Long silence occurs because of holding the line for calling other people, and listening to the called party. Long burst occurs because of background noise since voice gateway cannot distinguish between the actual talkspurt and the background noise. In [15],aggregateVoIP models are analysed.They model the VoIP as two state Markov-Modulated Poisson Process (ON and OFF).

Then, we investigate how QoS of VoIP is evaluated in other networks. In [16], authors collect real VoIP packets via different clients, analyzed these packets and investigate on performance metrics in VoIP in LAN. They try to estimate the obtained QoS at the end user, which is known as Mean Opinion Score (MOS). We are inspired by this research and we try to find the MOS value at CRN networks to supply a QoS to SU.

In [17],QoS for VoIP performance is studied in WLAN. In this research, they collect real time data from a university and analysed handover effect in VoIP. Handover is used for VoIP clients changing from one Access Point (AP) to another AP in WLAN because of the mobility. This brings the idea that this research can also be applicable to CRN. In CRN, handover term can be used as SU moving from one channel to another.

In [18], authors analyse QoS of VoIP with different codec and different packet sizes. This research also leads us to implement these different codec and packet sizes in CRN.

3. VOICE OVER IP

Voice over IP (VoIP) is the growing technology that allows voice conversations to be carried over the Internet Network [17]. VoIP uses Session Initiation Protocol (SIP) [19] or H323 [20] for signaling.

3.1 Working Principle

Voice conversations have a "sender" and "receiver" roles. Sender and receiver change the roles in different portions of conversation.

Sender creates analog signal on the conversation. These analog signals are converted to digitized form via encoder [18]. At the receiver, incoming packets are put in to a playback buffer to overcome late received or non-received packets. These packets are decoded using identical decoder. The digital bit stream converted back into an analog signal and send to receiver.

A delay occurs untila packet is generated at sender and received at receiver. Figure 3.1illustrates end-to-end delay components in voice packets [18].

Using the notation listed below,

- $D_{end\text{-}to\text{-}end}$: End-to-end delay
- D_{enc}: Encoding delay
- D_{pack}: Packetization delay
- f: frame size
- l: the look-ahead delay
- p_e: processing delay

end-to-end delay is calculated with **(3.1)**

$$D_{end\text{-}to\text{-}end} = D_{enc} + D_{pack} \quad \textbf{(3.1)}$$

$$D_{enc} = f + l + p_e \quad \textbf{(3.2)}$$

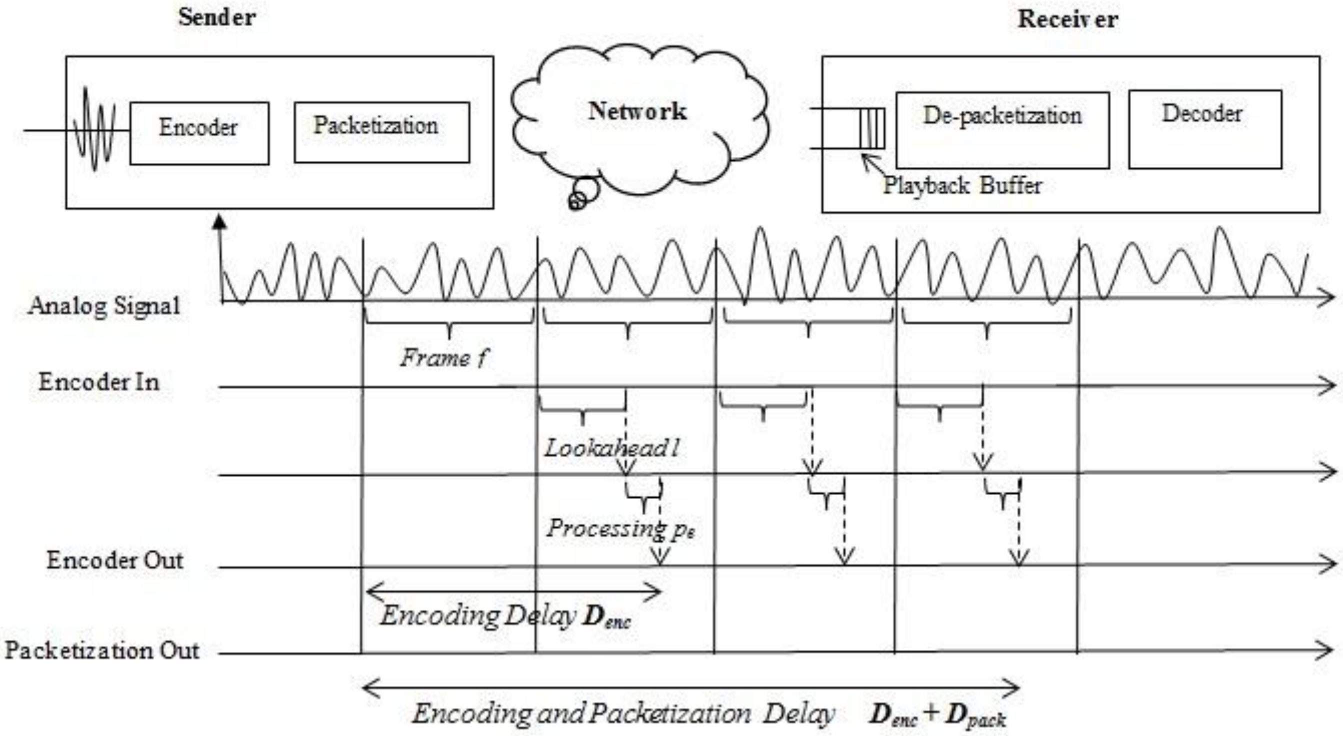


Figure 3.1 : End-to-end delay components for voice traffic [18]

D_{enc} changes according to encoding technique, which is the used codec in VoIP calls. This encoded packet is then packetized with including packetization delay D_{pack}. D_{pack} is the function of the number of frames k included in one packet [18].

There is a contrariety between rate of the encoded streams and the frame size, look-ahead delay, and processing delay. Sampling rate of the analog signal is important. If sampling rate of the encoded bit-stream r is low, then, the frame size, look ahead-delay, and processing delay gets larger. If sampling rate of the encoded bit-stream r is high, then, the frame size, look-ahead delay, and processing delay decreases.

3.2 Session Description Protocol

When initiating voice-over-IP calls, multimedia teleconferences, streaming video, or other sessions, all participants must use same media details, transport addresses, and other session description metadata. A standard representation of this information is introduced with [21].

SDP is independent of used transport and SDP message consists of three main sections: Session, Timing, and Media. Figure 3.2shows generic format of SDP. The one with * are optional.

Used SDP attributes in this thesis, rtpmap and ptime, are described in following sections. Other SDP header and attribute informationare detailed in Appendix B or more information can be retrieved from [21].

3.2.1 RTPMap

This attribute maps the RTP payload type number defined in "m=" to an encoding name, clock rate and encoding parameters. RTP payloads are defined by IANA and then standardized by IETF [22-23].

General format of rtpmap is:

a=rtpmap = <payload type><encoding name>/<clock rate> [/<encoding parameters>]

There are two different type RTP payloads: Static and Dynamic. If m line contains static assignments of payload, "a=rtpmap:" attribute may or may not be added according to client implementation. If RTP payload is dynamic, there must be an rtpmap attribute indicating the payload specific properties.

```
Session description
   v=  (protocol version)
   o=  (originator and session identifier)
   s=  (session name)
   i=* (session information)
   u=* (URI of description)
   e=* (email address)
   p=* (phone number)
   c=* (connection information -- not required if included in
        all media)
   b=* (zero or more bandwidth information lines)
   One or more time descriptions ("t=" and "r=" lines; see below)
   z=* (time zone adjustments)
   k=* (encryption key)
   a=* (zero or more session attribute lines)
   Zero or more media descriptions

Time description
   t=  (time the session is active)
   r=* (zero or more repeat times)

Media description, if present
   m=  (media name and transport address)
   i=* (media title)
   c=* (connection information -- optional if included at
        session level)
   b=* (zero or more bandwidth information lines)
   k=* (encryption key)
   a=* (zero or more media attribute lines)
```

Figure 3.2 : SDP format – Adapted from [21]

For example:

- Static payload*:*

  m=audio 49232 RTP/AVP 0

  "0" represents u-law PCM coded single-channel audio sampled at 8 kHz.

- Dynamic Payload:

  m=audio 49232 RTP/AVP 98

  a=rtpmap:98 L16/16000/2

  Dynamic RTP/AVP payload type 98 is given in m line. "rtpmap" header of this 98 is given as 16-bit linear encoded stereo audio, sampled at 16 kHz.

  Payload Number to Codec mapping can be reached fromTable A.1.

3.2.2 Ptime

Ptime is used for the length of time in milliseconds represented by the media in a packet. It is not necessary to know ptime to decode RTP.

General format of ptime is:

a=ptime:<packet time>

3.3 VoIP Session Initialization via SIP

VoIP subscribers can be registered from different clients. For example, at work they can be reached via soft client on work computer, or can be registered at mobile phone on the road, or registered from IP-phone at home, or can be registered from all of them at the same time.

Calling and called party does not know what the IP address of each other is. Clients talk to application server (AS) and get assigned services. In each new call, AS routes the call to appropriate IP of the called party.

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 via session initialization. Packet size is negotiated by codec; packet rate is negotiated by ptime value.

There are two different session initialization process in SIP: Fast Start, Slow Start [19]More detailed information about signals can be found at Appendix C.

3.3.1 Fast start session initialization

Calling party adds all the supported codecs to “m” line with requested order. However, called party decides which codec is going to be used in that particular call.

As illustrated in Figure 3.3, originator sends an INVITE message to AS with adding supported codecs to SDP. AS routes this INVITE to called user’s registered destinations. When called party answer the call, it chooses a codec from callers supported list and tells the chosen codec in 200 OK message, sends this 200 OK to AS. AS routes the 200 OK to calling party. Calling party sends ACK message to acknowledge the session is established. Then, AS forwards this ACK message to called party. After called party receives ACK, session is established. Users send the voice packet via agreed codec packet size and packet rate.

If calling party and called party do not have a common codec, than called party sends “488 Not Acceptable Here” to INVITE as shown in Figure 3.4, Calling and called party cannot establish a call until they have a common supported codec.

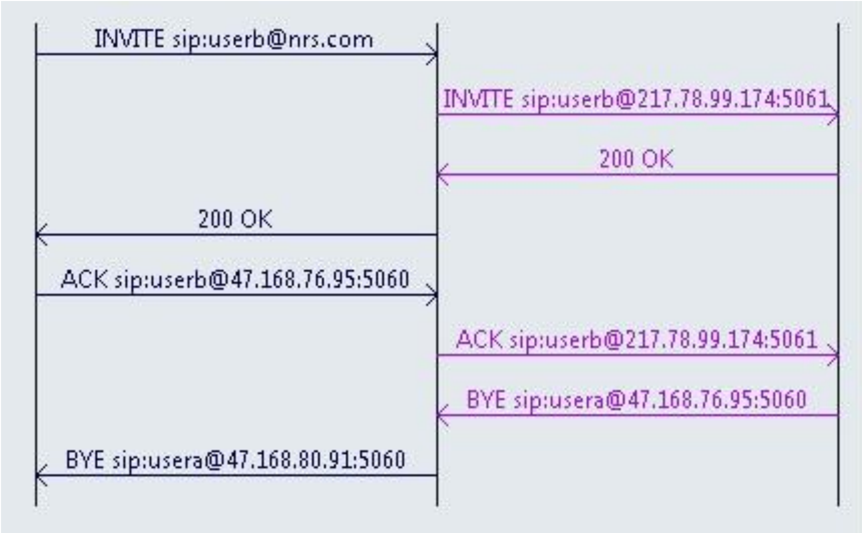


Figure 3.3 : Fast start session initialization

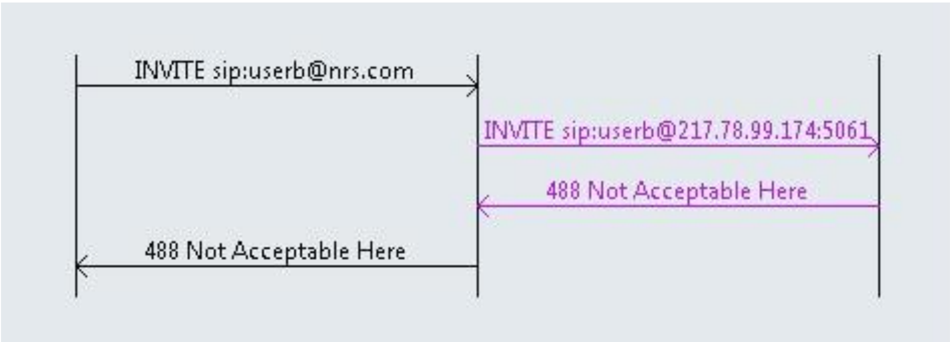


Figure 3.4 : Fast start - session initialization – no common codec

3.3.2 Slow start session initialization

Called party adds all the supported codecs to “m” line with requested order. However, calling party decides which codec is going to be used in that particular call. Figure 3.5 illustrates slow start session initialization signaling.

Originator sends an INVITE without SDP to AS. AS routes this INVITE to called user’s registered destination. When called party answer the call, it adds supported codecs to SDP portion of 200 OK message and sends this 200 OK to AS. AS routes the 200 OK to calling party. Originator choses one of the codecs in 200 OK and adds chosen codec to ACK message. AS routes this ACK to called party. After called party receives ACK, session is established. Users send the voice packet via agreed codec packet size and packet rate.

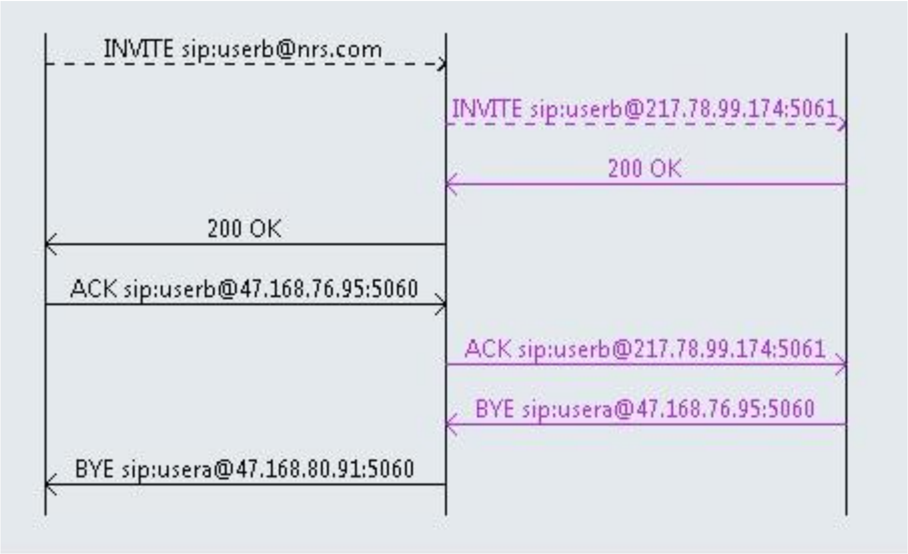


Figure 3.5 : Slow start session initialization

If calling party and called party do not have a common codec, than, calling party sends BYE to 200 OK instead of ACK. This signaling can be seen at Figure 3.6.

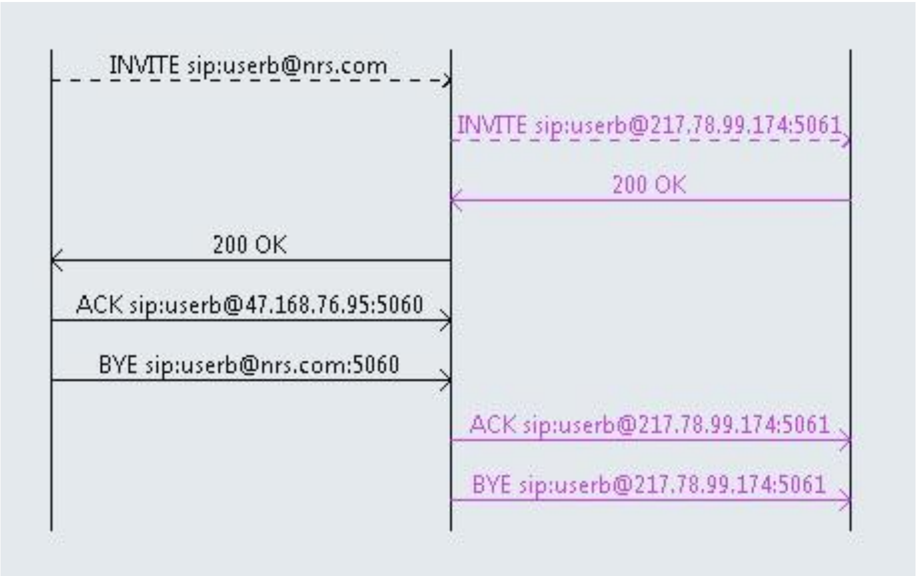


Figure 3.6 : Slow start - session initialization – no common codec

After signaling is established, calling and called party sends RTP packets in chosen codec's packet size, packet rate. When one of the parties would like to end the call, it sends BYE message to other party.

3.4 VoIP Codec Change

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

3.4.1 Codec change by update

After session is established, one of the parties sends an UPDATE (requestor) and holds the session with putting 0.0.0.0 to connection. Other party receiving UPDATE (responsal) sends a 200 OK. Then, requestor sends another UPDATE with changing the RTP Payloads of "m" line. Responsal sends 200 OK to UPDATE. Finally, codecs have been changed. This signaling is illustrated in Figure 3.7.

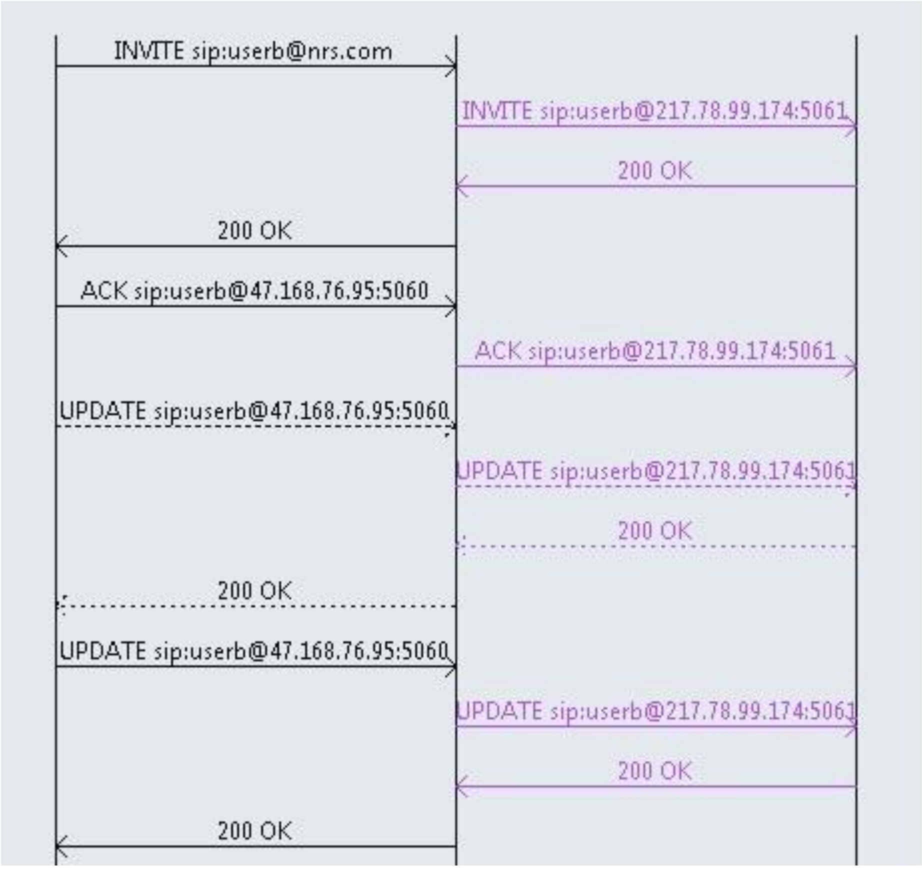


Figure 3.7 : Codec change via UPDATE

3.4.2 Codec change by invite

After session is established, one of the parties sends an INVITE (requestor) and holds the session with putting 0.0.0.0 to connection. Other party receiving INVITE (responsal) sends a 200 OK. Requestor sends ACK to 200 OK. Then requestor sends another INVITE with changing the RTP Payloads of "m" line. Responsal sends 200 OK to INVITE. Requestor sends ACK to 200 OK. Finally, codecs are changed. This signaling is illustrated in Figure 3.8.

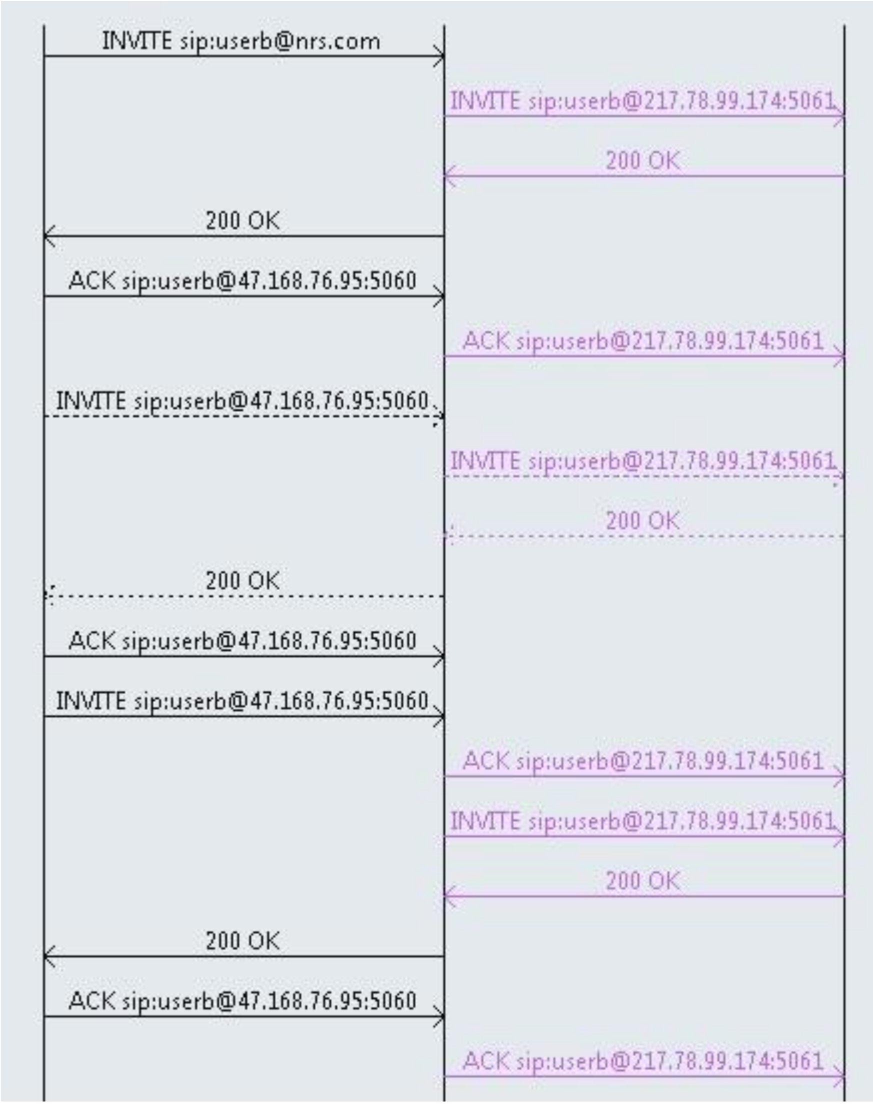


Figure 3.8 : Codec change via INVITE

3.5 RTP Packet and Codec Relations

An RTP packet consists of 20 bytes IP header, 8 bytes UDP header, 12 bytes RTP header and a variable size payload according to used codec as illustrated inFigure 3.9 [24].

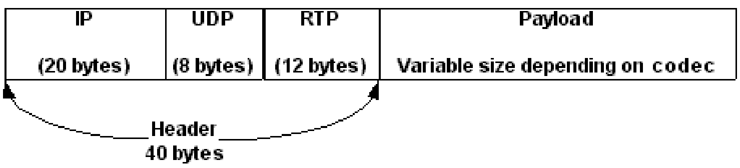


Figure 3.9 : RTP packet– Adapted from [24]

Popular used codecs in VoIP have been identified.

Using notations below,

- ***Codec Bit Rate (Kbps) (CBR):*** Based on the codec, this is the number of bits per second that need to be transmitted to deliver a voice call.
- ***Codec Sample Size (Bytes) (CSS):*** Based on the codec, this is the number of bytes captured by the Digital Signal Processor (DSP) at each codec sample interval.
- ***Codec Sample Interval (ms) (CSI):*** This is the sample interval that codec operates.
- ***Voice Payload Size (Bytes) (VPS):*** The voice payload size represents the number of bytes (or bits) that are filled into a packet. The voice payload size must be a multiple of the codec sample size. For example, G.729 codec sample size is 10, so, G729 packets can use 10, 20, 30, 40, 50, or 60 bytes of voice payload size.
- ***Voice Payload Size (ms):*** The voice payload size can also be represented in terms of the codec samples.
- ***Packets Per Second (PPS):*** PPS represents the number of packets that need to be transmitted every second in order to deliver the codec bit rate

their packet size, packet interval has been calculated using formulas **(3.3) (3.4) (3.5).**

$$CBR = CCS / CSI \quad \textbf{(3.3)}$$

$$PPS = CBR / VPS \quad \textbf{(3.4)}$$

$$T_{PS} = L2header + (IP / UDP / RTPheader) + VPS \quad \textbf{(3.5)}$$

For example, calculation of G729 codec with 20 ms payload is given below. All results of different codecs can be retrieved fromTable 3.1.

G729 codec's CSI is 10 ms;CSS is 10 bytes [25]. L2 layer header is taken as 6 byte for CRN.

CBR =10 Bytes / 10 ms = 1 Bytes /ms = 8 Kbps

PPS = 8 Kbps /20 * 8 bit = 50

WhenCI is 10 ms and CSS is 10 byte, G729 codec with 20 ms payload will be 20 byte.

TPS = 6 + 40 + 20 = 66 bytes

Table 3.1 : Codecs packet size and packet intervals according to voice payload

Codec	Codec Bit Rate (Kbps)	Codec Sample Interval (ms)	Codec Sample Size (Bytes)	Voice Payload Size (ms)	Voice Payload Size (Bytes)	Packets Per Second (PPS)	Total packet Size (46 + VPS) (Bytes)	Packet Interval (ms)
G711	64	10	80	10	80	100	126	10
G711	64	10	80	20	160	50	206	20
G711	64	10	80	30	240	34	286	30
G729	8	10	10	10	10	100	56	10
G729	8	10	10	20	20	50	66	20
G729	8	10	10	30	30	33.3	76	30
G729	8	10	10	40	40	25	86	40
G729	8	10	10	50	50	20	96	50
G729	8	10	10	60	60	17	106	60
G 723.1	6.3	30	24	30	24	33.3	70	30
G 723.1	6.3	30	24	60	48	17	94	60
G726-32	32	5	20	20	80	50	126	20
G726-32	32	5	20	30	120	33.3	166	30
G726-32	32	5	20	40	160	25	206	40
G726 -24	24	5	15	20	60	50	106	20
G726 -24	24	5	15	30	90	33.3	136	30

3.6 QoS Evaluation in Voice Over IP

IP Networks were built for non-real 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 [26]. In VoIP, a conversation between at least two people is trying to be kept over distances. 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 increase in VoIP usage is relative to the quality of the conversations. Users are accommodated to traditional public switched telephone networks (PSTN) whose voice quality is relatively standard and predictable.It provides an optimal service for time-sensitive voice applications that require low delay, low jitter, and constant but, low bandwidth [24].

The quality of voice is subjective. Users express the quality of a call as “good”, “bad”, “quite good” or “very bad”. This expression changes according to user earlier experiences, environment of the user, the mood of the user etc. 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 the best. The numerical method of expressing voice and video quality is defined as Mean Opinion Score (MOS). [21-22]

Instead of subjective tests, MOS can be calculated by voice quality effecting factors.

Following factors affect the voice quality: [24][26-27]

- Bandwidth**:** The capacity of Internet connection is the most important factor. Quality has a similar change with the bandwidth. If bandwidth is small, then quality degrades. If there is a big bandwidth, quality increases.
- Delay**:** The total time that a packet is sent from originator received at terminator. Delay consists of processing delay and network delay. Processing delay is changed according to used codec
- Jitter**:** Jitter is used for the packet arrive variations in time. Voice packets can arrive un-ordered to receiver. Incoming packets are kept in a buffer to overcome this problem, and played to user with delay.
- Packet Loss: The amount of the packets, which has been dropped, cannot be received via receiver. Retransmission is not a solution in real time applications, because voice packets are time sensitive.
- Echo or Noisy Background: Echo and noisy places bring extra delays since these bad voices are also digitized, packetized and sent over the network.

From high level, packet loss, jitter can be under group of delay. The clarity depends on service layer parameters. Figure 3.10 gives the voice quality according to clarity, delay, and echo.

The ITU-T E-Model [28] defines an analytic model of voice quality between two connections known as "Voice Transmission Quality from Mouth to Ear". E-model calculates an “R” value, which is Rating Factor. R is relevant with equipment impairment factor method, and previous transmission rating models. The model estimates the conversational quality from mouth to ear as perceived by the user at the receive side, both as listener and talker [28].

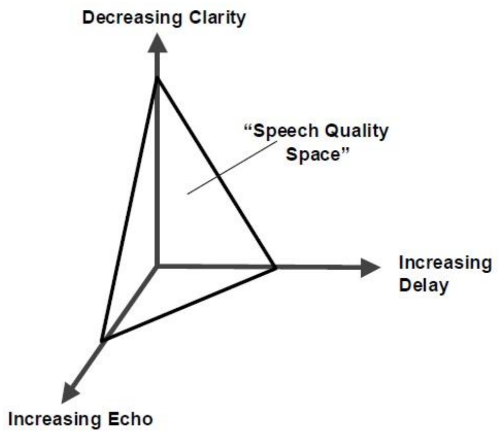


Figure 3.10 : Voice quality according to clarity, delay, and echo [26]

Using the notations listed below,

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

R-valueis calculated as **(3.6)** [28]:

$$R = R_o - I_s - I_d - I_{e-eff} + A \tag{3.6}$$

MOS value is calculated through R-value with **(3.7)** [29]

$$MOS = \left\{ \begin{array}{ll} R <= 6.5: & 1 \\ 6.5 < R <= 100: & 1 - \frac{7}{100}R + \frac{7}{6250}R^2 - \frac{7}{1000000}R^3 \\ R <= 100: & 4.5 \end{array} \right\} \tag{3.7}$$

The relation between R-value and MOS is referenced in Table 3.2 [28].

Table 3.2 : Relation between R value and MOS - Adoptedfrom [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

4. PROPOSED SOLUTION

After literature search in CRN, we have investigated VoIP working principle, and VoIP signaling details. Then, VoIP packets from Genband Application Server [30] in IP network of Netaş [31] had been collected and analyzed. Finally, a solution on the application layer is proposed to increase the QoS in VoIP application at SUs of CRN. VoIP working principle and VoIP signaling details have been given in earlier Section 3 - VOICE OVER IP.

4.1 Analyzed VoIP Scenarios

Generally, we tested basic call scenario with different codecs and different ptimes. There are two users: “User A (caller)” and “UserB (callee)”. Usersare registered to Genband Application Server through Netaş IP Network. User A calls User B. They talk approximately 1 min, and close the session. *Ethereal*is used to collect voice packets on both caller and callee with different codecs and different ptimes. Inspected codecs and ptimes are:

- G711 with ptime = 20 ms, 30 ms, 60 ms
- G729 with ptime = 20 ms, 30 ms, 60 ms
- PCMU with ptime = 20 ms, 30 ms, 60 ms

4.1.1 Analysis outcomes

Overall traffic of a basic call with using G729 Codec with ptime 60 ms is given in Figure 4.1.

SIP signaling in this traffic is seperated in Figure 4.2. As seen in the figure, SIP traffic happens in the beginning of call and at the end of the call. This is expected, because, calling and caller deals on SDP session parameters at the beginning of the call, and close the media session at the end of the call. These are done via INVITE – 200 OK – ACK SIP requests at the beginning and BYE at the end of the call.

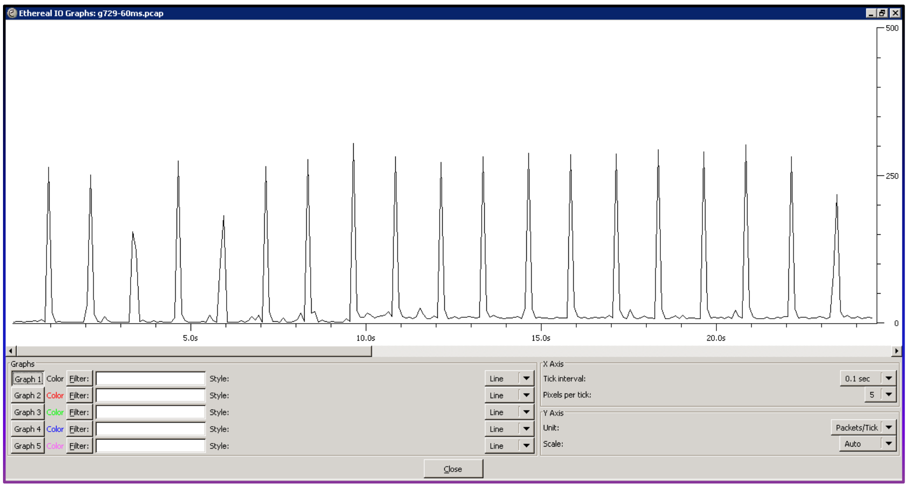


Figure 4.1 : Overall traffic with G729 – ptime 60 ms

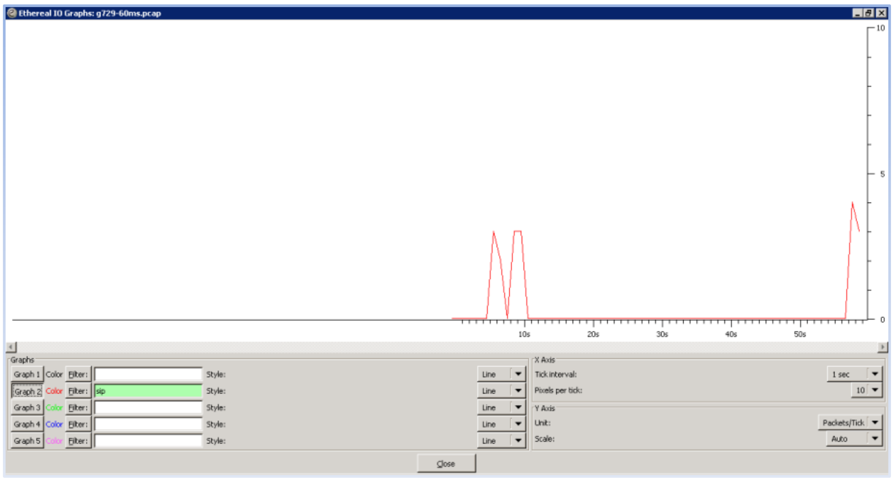


Figure 4.2 : SIP Traffic with G729 – ptime 60 ms

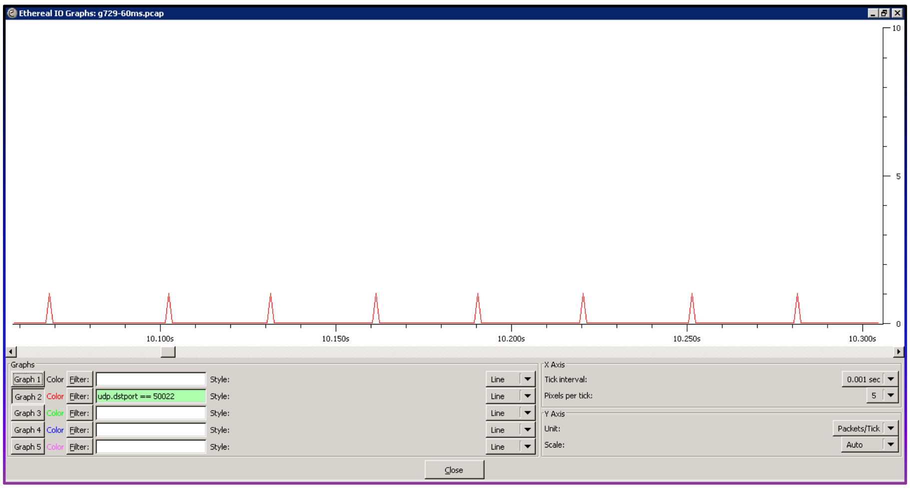


Figure 4.3 : VoIP packets when codec is G729, ptime is 60

Figure 4.3 andFigure 4.4 shows the packet rate when different ptime is used.

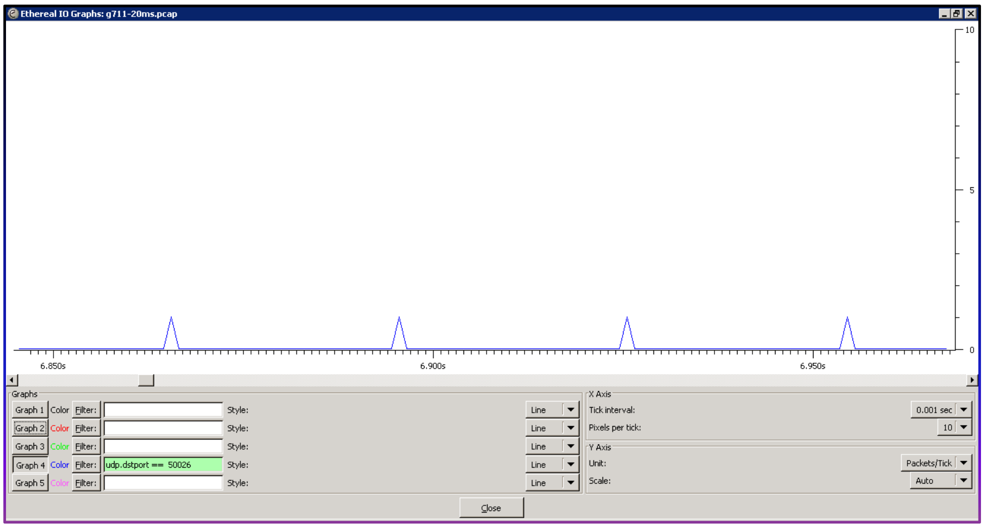


Figure 4.4 : VoIP packets when codec is G729, ptime is 20

4.2 Proposed Solution to Enhance QoS in SU of CRN

After analysis on collected packets, it is noticed that VoIP packets packet size is constant and these packets are generated periodically. According to SIP RFCs and experimental data shows that packet sizes are relevant to negotiated codec that is given in “rtpmap” attribute at session initialization. Packet generation interval is related to “ptime” attribute. That’s why, VoIP traffic can be modeled as Constant Bit Rate (CBR) traffic.

The quality of a VoIP call in CRN can decrease or increase in time relative to PU arrivals, other SU channel usage, etc. According to analysis on collected packets, literature reviews and VoIP standards, we proposed an algorithmon application layer to supply a satisfactory QoS level to SU employing VoIP applications as given in Table 4.1.

Our proposition isto calculate the QoS of a VoIP call periodically. In this way, we can notice QoS decreases, manipulate the call, and increase the QoS again. As told in Section 3.6, QoS of a VoIP call is calculated through MOS value. MOS value is relative to R-value as given earlier in Table 3.2.

In proposed algorithm,R-valueis calculatedperiodically according to delay, jitter and packet loss. Algorithm reaches to MOS value from R-value.In each period, algorithm compares the earlier period MOS value withcurrent MOS value. If difference is zero, it means that MOS value remained same. If difference is bigger than zero, it means that MOS value increased. If difference is less than zero, it means that MOS value decreases. On here according to decrease rate, we propose to change ptime or codec value. As given in Section 3.4 - VoIP Codec Change, calling or caller can change the negotiated session parameters in call. When difference is one, we propose to increase ptime value. In this way, packet rate is decreased. Each packet carries more voice packet. This supplies SU to wait more time for non-received packets. If difference is more than one, then codec value is changed to a lower codec. In this way, total packet size is decreased. Table 4.1 gives a pseudo codeof ourproposal.

Table 4.1 : Pseudo code of proposed algorithm

```
Calculate delay, jitter and packet loss in each period
Calculate R value according to delay, jitter and packet loss
Calculate MOS according to R value and save it
Compare the earlier period MOS value and new MOS value
Set the difference between new MOS value and earlier MOS value
if(difference is equal to 0)
   # MOS value remained same
else if((difference is less than < 0)
   # MOS value decreased
   if((difference is equal to -1)
increase the ptime value
   else
change the current codec to a lower codec
else if(difference is bigger than 0)
    # MOS value increased
endif
```

R value calculation formula by ITU-T model was given in Section 3.6. However, this formula includes some attributes that we can’t calculate via simulation tool. That is why, we used algorithm in [32] to calculate R-value according to delay, jitter and packet loss. Table 4.2 gives the pseudo code of R value calculation that we used.

Table 4.2 : Pseudo code of R-value calculation – Adapted from [31]

```
Calculate the delay, jitter, and packet loss in each period
Set the EffectiveLatency as the addition of average delay, doublejitter and
10 for protocol latency issues

if(EffectiveLatency <160)
```

```
Divide EffectiveLatency to 40 and remove it from 93.2
    Set the result as R value
else
Subtract 120 from EffectiveLatency, divide it to 10, and remove it from
   Set the result as R value
endif

Multiply 2.5 with percentage of packet loss
Deduct this value from R value
Set the result as R value
```

After our first tests with this algorithm, we found that it has some bugs. Since we only observe MOS decreases, we still use lower codecs although there is no PU around. It brings the result thatour improvements related to number of codecs. When we only observe decreases, we can improve the QoS according to number of our supported codec and ptime values. If VoIP conversation goes longer, we can't increase QoS anymore. That iswhy; we improved our algorithm as also observing MOS increases.

In improved version, we save a boolean when MOS increases. In addition, a counter is added to count how many period we remained in same MOS after an increased operation. When MOS value is same, we check whether an increase happened earlier or not. If happened, we increase the counter and check whether we remained in same MOS value more than 3 periods. When we hit the counter, we go and change to a better codec, and decrease the ptime. When there is a MOS decrease, we still do the earlier algorithm, also set the increase boolean to false.Table 4.3 shows pseudo code of improved algorithm.

Table 4.3 : Pseudo code of improved algorithm

```
Initialize increased value as false
Initialize counter as zero
Calculate R value according to delay, jitter and packet loss
Calculate MOS according to R value and save it.
Compare the earlier period MOS value and new MOS value
Set the difference between new MOS value and earlier MOS value
if(difference is equal to 0)
    # MOS value remained same
    if (increase)
      Increase the counter one more
   if (count is bigger than 3)
```

```
        Decrease the ptime and change the codec
        Set the increased boolean to FALSE
        Reset the counter
endif
endif
else if(difference is less than 0)
   # MOS value decreased
  Set the increased boolean to FALSE
  Reset the counter
  if(difference is equal to -1)
     increase the ptime value
   else
     change the current codec to a lower codec
   endif
else if(difference is bigger than 0)
    # MOS value increased
Set the increased boolean to TRUE
endif
```

4.3 Simulation Environment Details

A tcl script is written at NS-2 [33] to prove the proposed algorithm. The test bed is created as 500x500-grid area, which simulates 500 m by 500 m square area. There is one SU peer in simulation. They are 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 PU is modeled by moving the SU to a place longer than antenna's range. When PU has gone, SU moved back to original place.

Two mobile nodesare created with following initial positions in XY:

- N0: (150, 150)
- N1: (300, 300)

Only one channel is assigned to users since we thought that in a real time application, both of the agents (caller and called party) channel movement at the same time is not realistic. An RTP Agent is attached to first node N0. VoIP application is simulated with CBR application running over an RTP Agent. A Null receiver is attached to N1. RTP Agent and receiver are connected to each other. TwoWayGround Antenna, which has 250m range, is used. Calculated codec packetSize and interval rates (Table

3.1) are put into an array in tcl script. They are used hardcoded. Output files are analyzed via *awk*. Graphs are drawn via *gnuplot*.

In this paper, Cognitive Radio's Spectrum Sensing is out of scope. It is assumed that Spectrum Sensing is done; sensing units specify available channels and slots.

4.4 Simulation Results

Simulationis run for 20s. Tests run for two different PU traffic model: Deterministic and Stochastic.

4.4.1 Deterministic PU traffic models

The simulations are run under three different PU traffic models as listed below:

- ON Period = OFF Period
- ON Period < OFF Period
- ON Period > OFF Period
  - ON Period =1.5*OFF
  - ON Period =2*OFF
  - ON Period =3*OFF

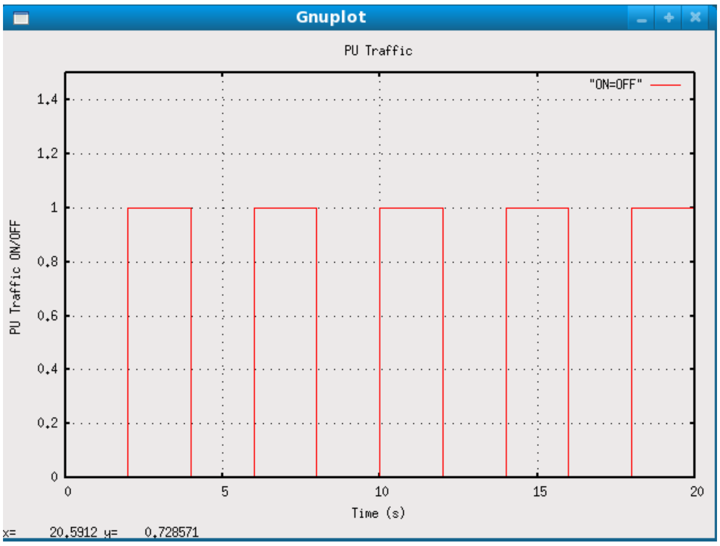
4.4.1.1 On period = off period

First, the technique is tested when PU ON and OFF periods are equal. ON and OFF periods are taken as 2s. While Figure 4.5(a) illustrates PU traffic model, Figure 4.5 (b) is MOS value change of the SU, with our algorithm.

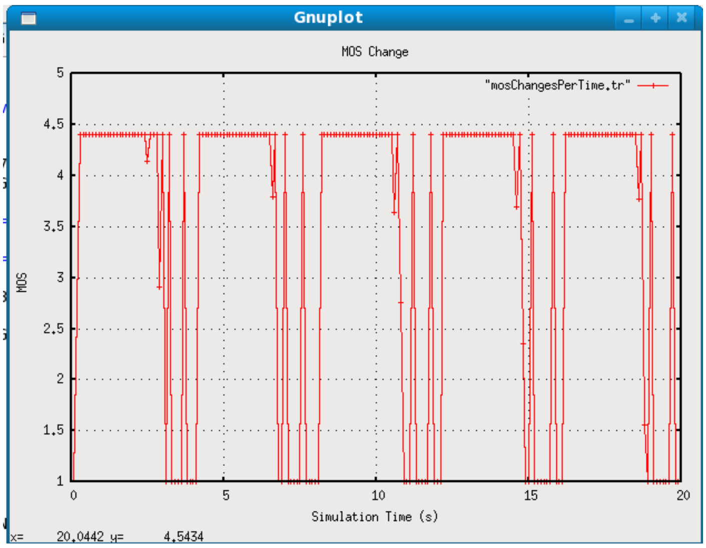
As seen from the figure, MOS starts to decrease at time 2.0, which is the PU arrival time. Since the algorithm notices a small decrease in MOS, it increases ptime. That's why MOS increases in the next MOS calculation period. Similar behavior occurs at time 2.8. When there is a bigger decrease than 1,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

4.4.1.2 On period < off period

Later, another traffic model, ON period of PU is less than OFF period, is tested. Figure 4.6 (a) illustrates PU traffic model. ON periods are taken as 2 s, while OFF periods are 3 s. Figure 4.6 (b) shows the MOS changes in SU according to the behavior of the PU. Changes in MOS start to decrease at time 8.0, 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. When there is a greater decrease like at time 9.0, algorithm changes codec and in the next period, MOS increases.

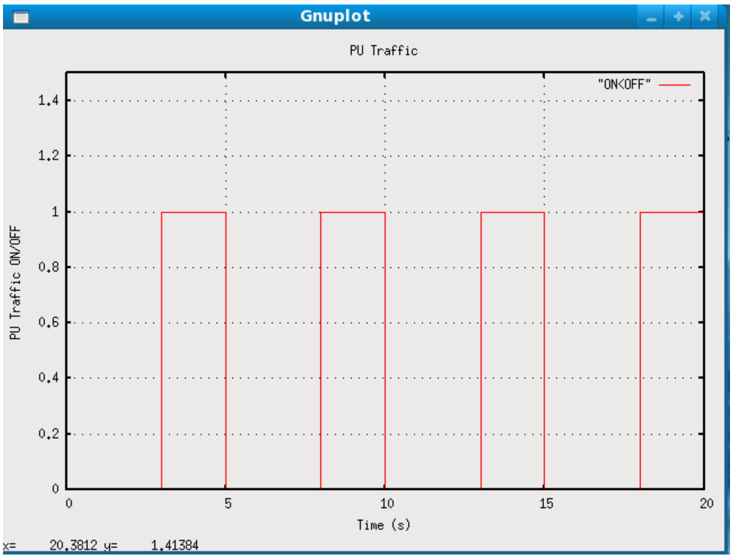


(a)

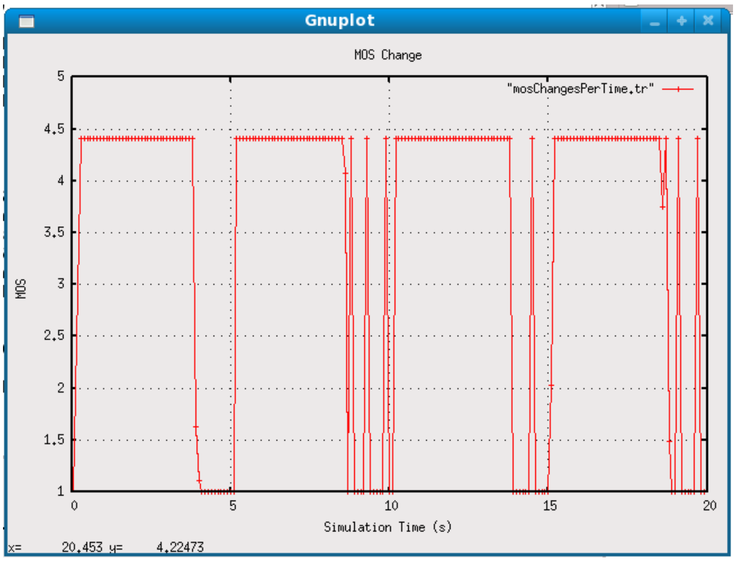


(b)

Figure 4.5 : ON Period = OFF Period:(a) PU traffic (b) MOS change on SU



(a)



(b)

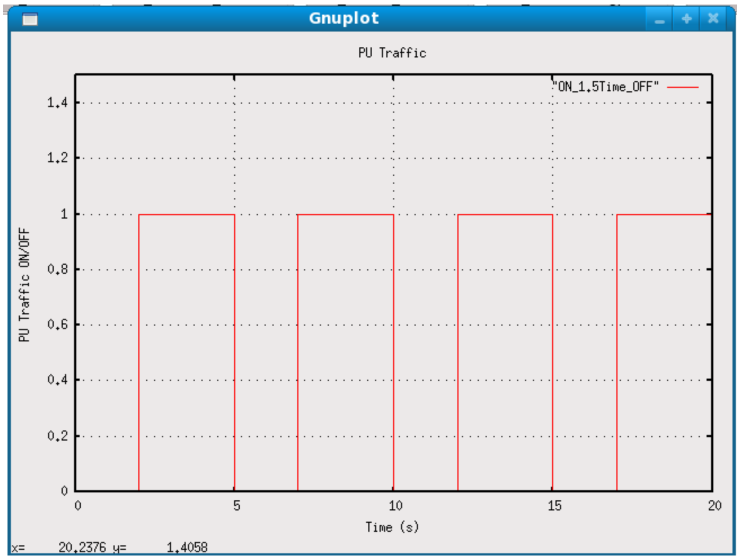
Figure 4.6 : ON Period < OFF Period: (a) PU traffic (b) MOS change on SU

4.4.1.3 On period > off period

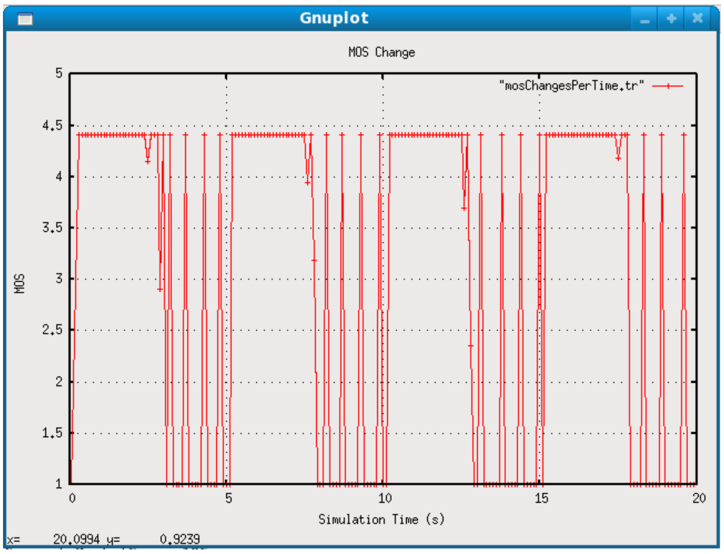
Lately, algorithm is tested on traffic model where ON period is bigger than OFF period. Since the bigger rate is important, three different rates are tested.

4.4.1.3.1 On period= 1.5*off period

In this test, ON period is taken as 1.5 times of OFF period. Figure 4.7(a) shows the PU traffic. OFF period is 2s, ON period is 3 s. Figure 4.7(b) shows MOS value change of the SU VoIP application with our algorithm. As seen from graphic, algorithm can handle this type of traffic. MOS starts to decrease at time 2.0, which is the PU arrival time. Since the algorithm notices a small decrease in MOS, it increases ptime. That's why MOS increases in the next MOS calculation period. Similar behavior occurs at time 7 s. When there is a bigger decrease than 1, like at time 4.0, the algorithm changes codec and in the next period MOS increases again.



(a)

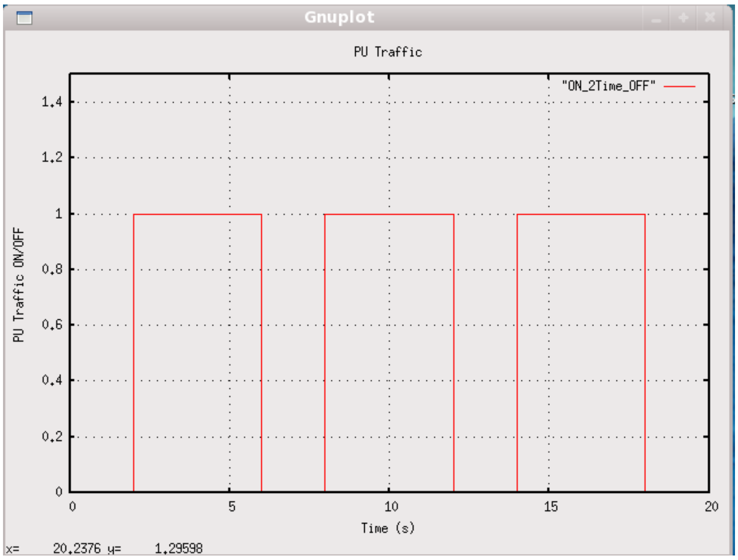


(b)

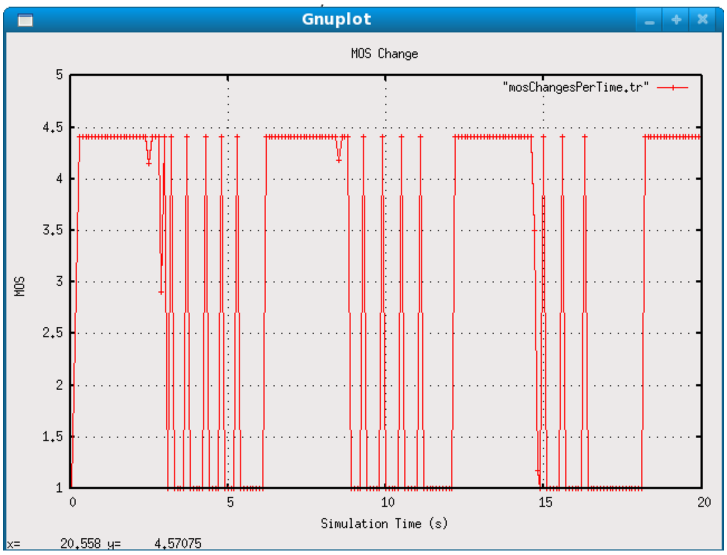
Figure 4.7 : ON Period =1.5*OFF Period: (a) PU traffic (b) MOS change on SU

4.4.1.3.2 On period = 2*off period

In this test, ON period is taken as 2 times of OFF period. Figure 4.8 (a) shows the PU traffic, OFF period is 2 s, ON period is 4 s. Figure 4.8 (b) shows MOS value change of the SU VoIP application with our algorithm. As seen from graphic, algorithm can still handle this type of traffic. MOS starts to decrease at time 2.0, which is the PU arrival time. Since the algorithm notices a small decrease in MOS, it increases ptime. That's why MOS increases in the next MOS calculation period. When there is a bigger decrease than 1, like at time 3.0, the algorithm changes codec and in the next period MOS increases again.



(a)

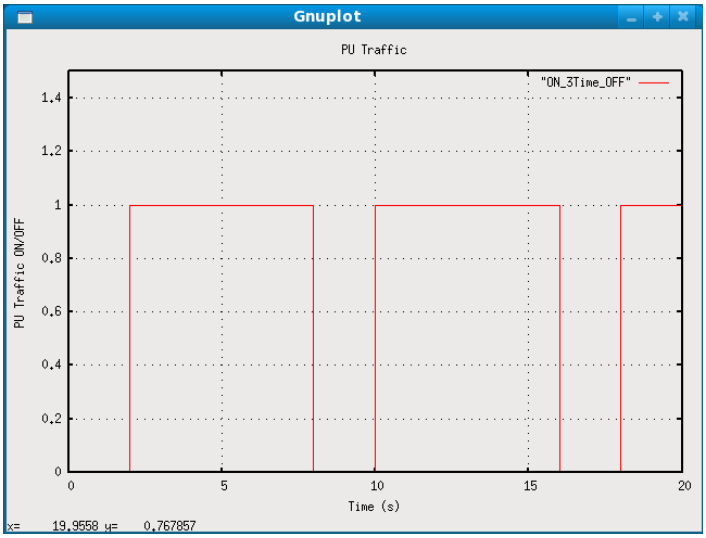


(b)

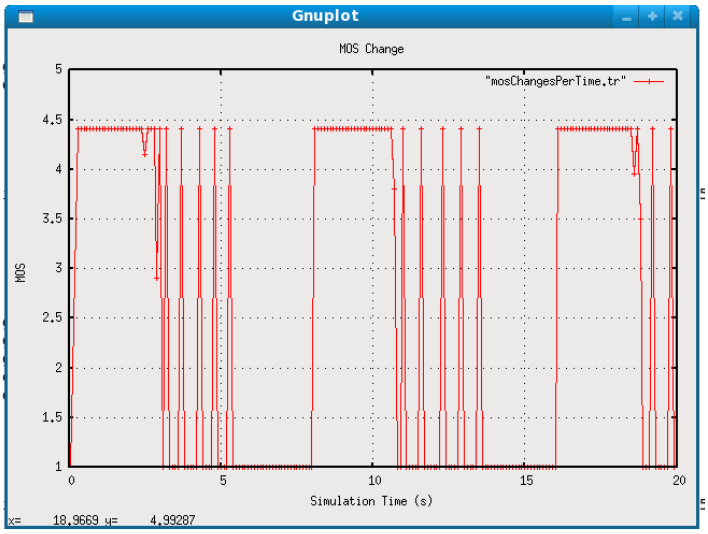
Figure 4.8 : ON Period =2*OFF Period: (a) PU traffic (b) MOS change on SU

4.4.1.3.3 On period = 3*off period

In this test, ON period is taken as 3 times of OFF period. Figure 4.9 (a) shows the PU traffic, OFF period is 2 s, ON period is 6 s.Figure 4.9 (b) shows MOS value change of the SU VoIP application with our algorithm. As seen from graphic, algorithm cannot increase the MOS value after 6 s. This is because of codecs and ptimes that has been known by algorithm. Algorithm will try to change the codec according to the ones in its database. Although, it starts to change codecs in big MOS decreases, after a while it doesn't have any other codec to be able to change. That's why after 6 s, algorithm don't have any remained codec, so QoS is down until PU leaves the channel. This can be prevented by supporting more codec by SU clients.



(a)



(b)

Figure 4.9 : ON Period =3*OFF Period: (a) PU traffic (b) MOS change on SU

Figure 4.10shows the mean MOS values for these traffic types.

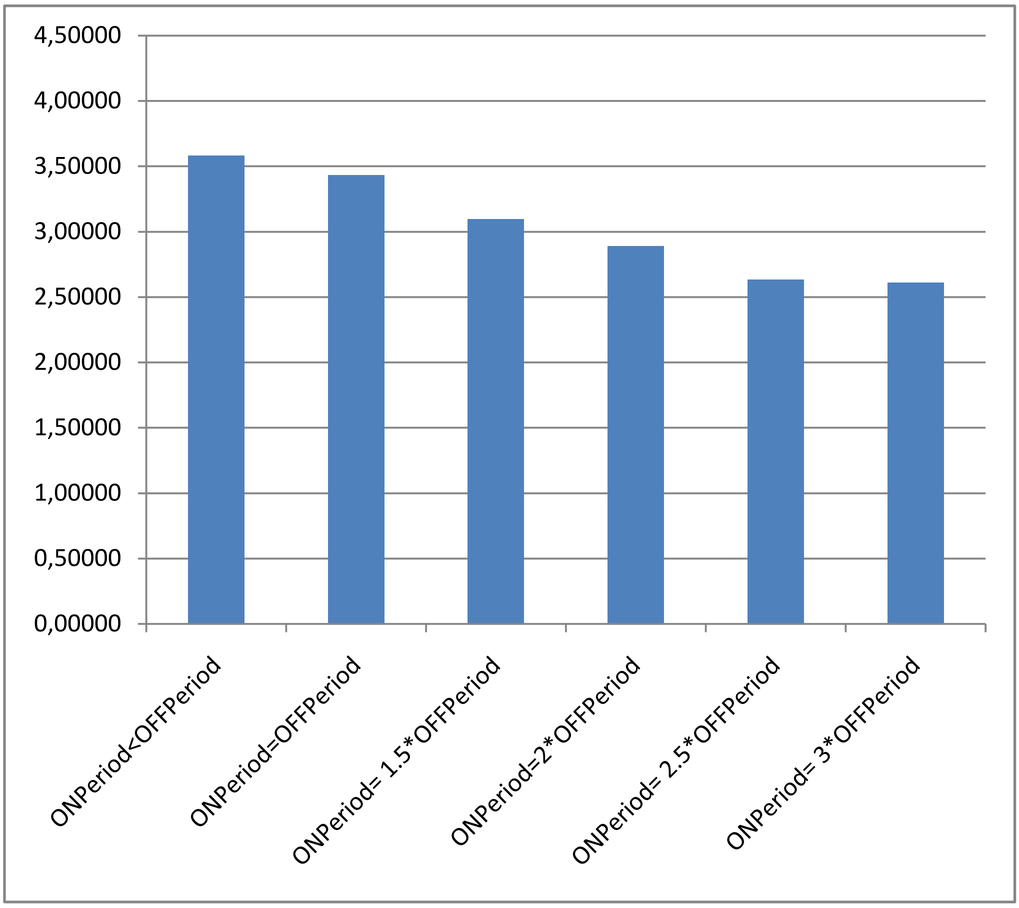


Figure 4.10 : Mean MOS value for deterministic traffic models

4.4.2 Stochastic PU traffic models

After proving our algorithm results with deterministic PU traffic model, we tested the algorithm with stochastic PU models. For that purpose, we generated random traffic between0 and 10with ns-random facilities. After generating random traffic, we defined whether PU is on the channel or not via Energy Detector. If PU SNR rate is bigger than 5, we said that PU is ON the channel.

We run the test on 10 different traffic models.Figure 4.11is an example for the generated traffic with NS-2 for PU.

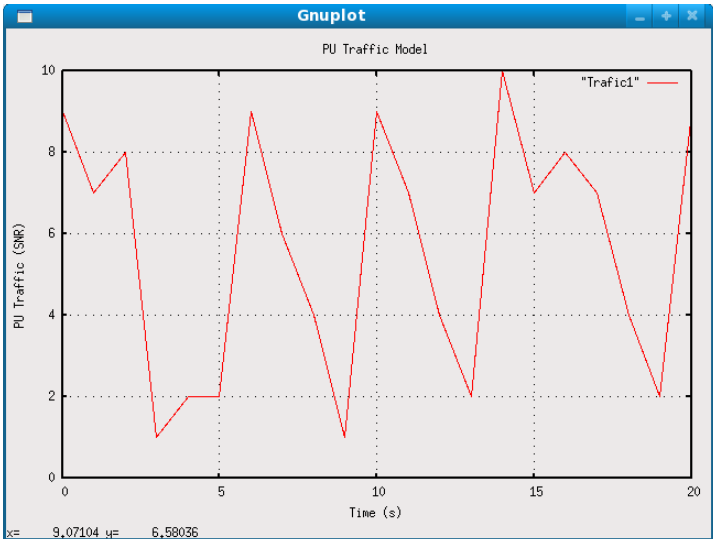


Figure 4.11 : Random PU traffic

Figure 4.12 shows how PU traffic is changed to ON/OFF traffic via energy detector.

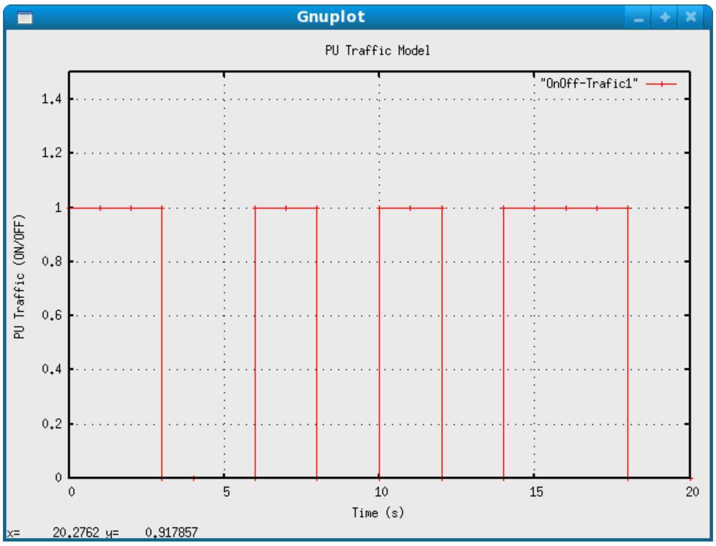


Figure 4.12 : PU model of Figure 4.11

Figure 4.13 shows the MOS changes in current SU with proposed algorithm. As seen in graph, algorithm shows good results also on non-deterministic traffics.

Other traffic results can be investigated from Appendix D.

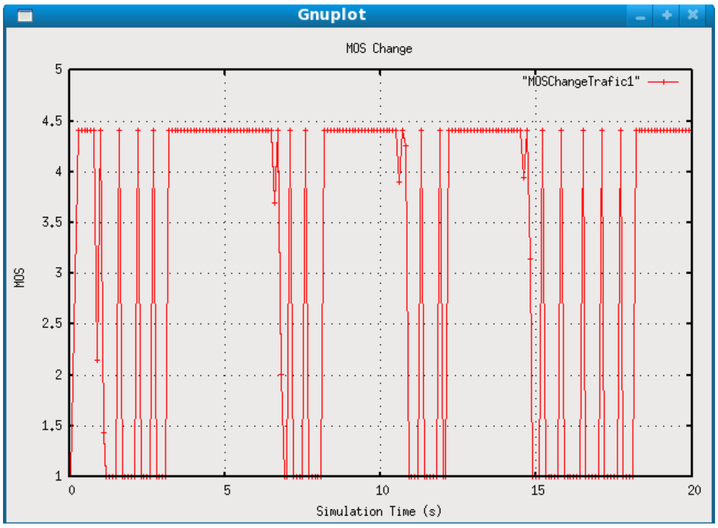


Figure 4.13 : MOS change on random PU traffic

Figure 4.14 shows the mean MOS value for these 10 traffic types.

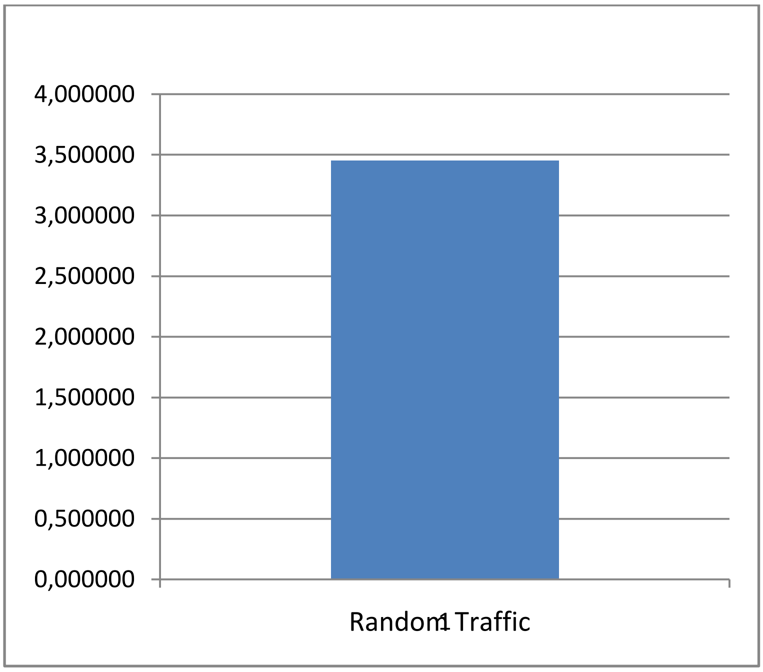


Figure 4.14 : Mean MOS of 10 random PU traffic

4.4.3 Negative exponential traffic models

After running stochastic PU traffic model, we tested the algorithm with PU arrivals and departures are Negative Exponential Function.

Using the notations listed below,

- *a:* Average
- *F(x)* = Negative Exponential Function

$$F(x) = 1 - e^{-x/a} \quad \textbf{(4.1)}$$

$$F(x) = r \rightarrow r = 1 - e^{-x/a} \quad \textbf{(4.2)}$$

$$\ln r = -\frac{x}{a} \ln e \quad \textbf{(4.3)}$$

$$x = -a \ln(r) \quad \textbf{(4.4)}$$

x value is calculated through **(4.1), (4.2), (4.3), (4.4).**

For that purpose, we generated random traffic between 0 and 1 for r value with ns-random facilities. Then, we calculated the x value from r value according to formula given in **(4.4).**These x values are taken as PU ON and PU OFF models. Algorithm is tested under 10 different random traffic models in each average rates given below:

- Average ON rate = Average OFF rate
- Average ON rate < Average OFF rate
- Average ON rate > Average OFF rate

All traffic results can be investigated from Appendix E.

According to traffic results, algorithm maintains MOS value. This way, it provides a QoS level to SU. Traffic results on “Average ON rate > Average OFF rate” show that algorithm cannot increase MOS value after some time. This is related to hardcoded codec and ptime attribute size. If there is more codec available, algorithm can maintain MOS value.

Figure 4.15 shows the mean MOS values in 10 traffic model for each Negative Exponential traffic.

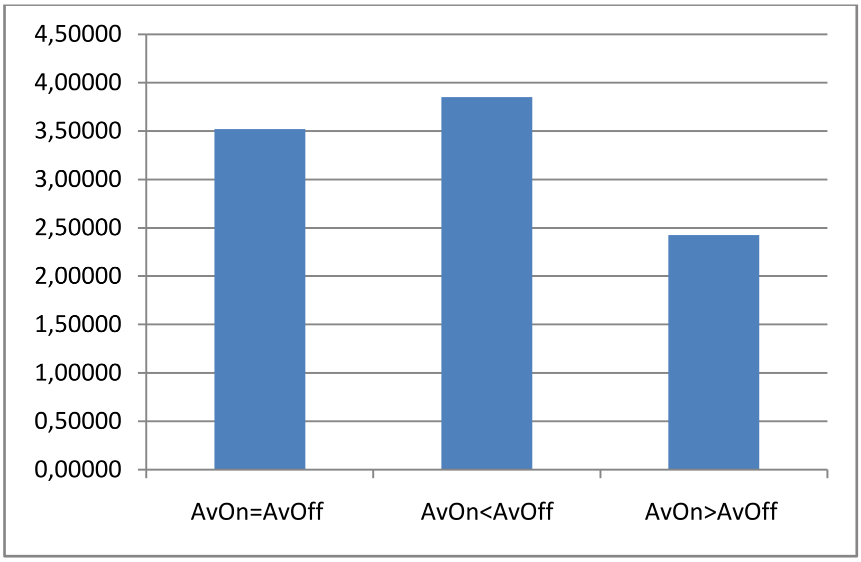


Figure 4.15 : Mean MOS value for negative exponential PU traffic models

5. CONCLUSIONS AND RECOMMENDATIONS

The interest in Cognitive Radio Networks is growing rapidly because of CRN improvements on spectrum usage. Until now, research works were on Cognitive Radio's working principles; like spectrum sensing, spectrum sharing etc. Since these working principles start to be settling down, QoS is becoming a major research area in CRN.

Current research works on QoS in CRN are categorized in two groups: QoS of Primary Users, and QoS of Secondary Users. First category researchers are proving that existing QoS of PU not degrades by SU usage in PU licensed bands. The latteris working on providing QoS level to SU.

In this study, Quality of Service to Secondary Users is studied. One of the most popular applications Voice over IP is chosen. Providing a satisfying QoS level to SUisaimed. For this purpose, real VoIP packets collected from Genband Application Server, and analyzed. With the help of analyzing VoIP packets and VoIP signaling, a solution in application layer is proposed to provide a QoS to SU. Then, we simulated our proposal at NS-2. Our proposal obtains proving results fromNS-2 simulations.

If cognitive radio devices implement the algorithm defined in this research, they can be aware of QoS decreases and increase the QoS again by changing session credentials.

For future work, other applications such as video calls, data transfers can be investigated.

A part of this study got approval and presented in ICSNC 2011, October 23-28, 2011- Barcelona, Spain.

REFERENCES

[1]**Waze, V. L.** (2003).Spectrum Access and the Promise of Cognitive Radio Technology.*Cognitive Radio Workshop Presentations*, Washington, DC, May 19.

[2] **Wang, B., & Liu, K. J. R.** (2011).Advances in Cognitive Radio Networks: A Survey.*IEEE Journal of Selected Topics in Signal Processing*, *5(1)*,5-23.

[3] **Akyildiz, I. F., Lee, W. Y., Vuran, M. C., &Mohanty, S.**(2006).NeXt generation/dynamic spectrum access/cognitive radio wireless networks: A survey.*Computer Networks*,*50(13)*, 2127–2159.

[4] **Haykin, S.**(2005). Cognitive radio: brain-empowered wireless communications.*Selected Areas in Communications, IEEE Journal on*, *23(2)*, 201–220.

[5] **Akyildiz, I. F., Lo, B. F., &Balakrishnan, R.**(2011).Cooperative spectrum sensing in cognitive radio networks: A survey.*Physical Communication*, *4*, 40-62.

[6] **Wu, Y., Tsang, &D. H. K.**(2009).Distributed Power Allocation Algorithm for Spectrum Sharing Cognitive Radio Networks with QoS Guarantee.*IEEE INFOCOM*, Rio de Janeiro, Brazil, April 19-25.

[7] **Yau, A. K. L.,Komisarczuk, P., &Teal., P. D.** (2008).C2net: A Cross-Layer Quality of Service (QoS) Architecture for Cognitive Wireless Ad Hoc Networks.*ATNAC*, Adelaide, SA, December 7-10.

[8] **Lien, S. Y., Prasad, N. R., Chen, K. C., & Su, C. W.**(2009). Providing Statistical Quality-of-Service Guarantees in Cognitive Radio Networks with Cooperation. *IEEE CogART*, Aalborg, Denmark, May 18-20.

[9] **Liang, Z., &Zhao, D.** (2010).Quality of Service Performance of a Cognitive Radio Sensor Network. *IEEE ICC 2010*, Cape Town, South Africa, May 23-27.

[10] **Wysocki, T., &Jamalipour, A.** (2010).Mean-Variance Based QoS Management in Cognitive Radio. *IEEE Transactions on Wireless Communications*,*9(10)*, 3281-3289.

[11] **Lee, H., & Cho, D. (**2010).Capacity Improvement and Analysis of VOIP Service in a Cognitive Radio System. *IEEE Transactions on Vehicular Technology*, *59(4)*,1646-1651.

[12] **Wang, P., Niyato, D., & Jiang, H.**(2010).Voice-Service Capacity Analysis for Cognitive Radio Networks.*IEEE Transactions on Vehicular Technology*, *59(4)*,1779-1790.

[13] **ITU-T Recommendation.**(1993). *P-59 - Telephone Transmission Quality Objective Measuring Apparatus Artificial Conversational Speech.*

[14] **Pragtong, P., Erke, T. J., & Ahmed, K. M.** (2005).Analysis and Modeling of VoIP Conversation Traffic in the Real Network. *ICICS 2005,* Bangkok Town, Thailand, December 6-9.

[15] **Hassan, H., Garcia, J.M. &Bockstal.C.** (2006).Aggregate Traffic Models for VoIP Applications.*Digital Telecommunications ICDT'06 International Conference,* Cap Esterel, Côte d'Azur, France, August 29 – 31.

[16] **Dimova, R., Georgiev, G., &Stanchev, Z.**(2008).Performance Analysis of QoS Parameters for Voice over IP Applications in a LAN Segment.*International Scientific Conference Computer Science,* AGH University of Science and Technology, Kraków, POLAND, June23-25.

[17] **Lakas, A., & Boulmalf, M.** (2007).Experimental Analysis of VoIP over Wireless Local Area Networks. *Journal of Communications, 2(4),* 3-9.

[18] **Cruz, H. T., & Torres-Román, D.** (2005).Traffic Analysis for IP Telephony.*2nd International Conference on Electrical and Electronics Engineering (ICEEE) and XI Conference on Electrical Engineering,* Amsterdam, the Netherlands, June 8-12.

[19]**Rosenberg, J.** (2002). SIP: Session Initiation Protocol. *Internet Engineering Task Force,* RFC3261.

[20] **ITU-T Recommendation.**(2006). *H.323.*

[21] **Handly, M., Jacabson, V., & Perkins, C.** (2006).*SDP: Session Description Protocol.*

[22] **IANA.(2011).***Real-Time Transport Protocol (RTP) Parameters*. Retrieved May 05, 2011 from http://www.iana.org/assignments/rtp-parameters

[23] **Schulzrinne, H., &Casner, S.** (2003). *RTP Profile for Audio and Video Conferences with Minimal Control.*

[24] **CISCO.** (2005).*Voice Over IP - Per Call Bandwidth Consumption.*

[25] **ITU-T Recommendation.**(2007). *G-729- Coding of speech at 8kbit/s using conjugate-structure algebraic-code-excited linear prediction.*

[26] **Pracht, S., & Hardman, D.** (Unknown). *Voice Quality in Converging Telephony and IP Networks.*

[27] **Unuth, N. (2010).** *Mean Opinion Score (MOS) - A Measure Of Voice Quality.*Retrieved May 05, 2011 from http://voip.about.com/od/voipbasics/a/MOS.htm

[28] **ITU-T Recommendation.**(2005).*G-107 - The E-model, a computational model for use in transmission planning.*

[29] **Möller, S., &Hoene, C.** (2002). Information about a new method for deriving the transmission rating factor R from MOS in closed form. *ITU-Telecommunication Standardization Sector.*

[30] **Genband.**(2011).*A2 Business Applications.* Retrieved May 05, 2011 from http://www.genband.com/Home/Products/Applications/A2-Business-Applications.aspx

[31] **NETAS. (2011).** *Netas.* Retrieved May 05, 2011 from http://www.netas.com.tr/

[32] **Nessoft. (2005).** *How is MOS calculated in PingPlotter Pro.* Retrieved May 05, 2011 from http://www.nessoft.com/kb/50

[33] **USC (University of South California). (2005).** *The Network Simulator - ns-2.* Retrieved May 05, 2011 from http://www.isi.edu/nsnam/ns/

APPENDICES

APPENDIX A : Payload Number to Encoding Information
APPENDIX B :SDP Details
APPENDIX C : Fast Start & Slow Start Session Initialization Messaging Details (In CD)
APPENDIX D : Random Traffics & Algorithm Results on These Traffics (In CD)
APPENDIX E: Negative Exponential Arrival and Departure Traffics & Algorithm Results on These Traffics (In CD)

APPENDIX A

Table A.1 : Payloadnumber to encoding information

Payload Number	Encoding name	Media Type	Clock Rate	Chan nel
0	PCMU	A	8000	1
1	reserved	A		
2	reserved	A		
3	GSM	A	8000	1
4	G723	A	8000	1
5	DVI4	A	8000	1
6	DVI4	A	16000	1
7	LPC	A	8000	1
8	PCMA	A	8000	1
9	G722	A	8000	1
10	L16	A	44100	2
11	L16	A	44100	1
12	QCELP	A	8000	1
13	CN	A	8000	1
14	MPA	A	90000	1
15	G728	A	8000	1
16	DVI4	A	11025	1
17	DVI4	A	22050	1
18	G729	A	8000	1
19	reserved	A		
20	unassigned	A		
21	unassigned	A		
22	unassigned	A		
23	unassigned	A		
24	unassigned	V		
25	CelB	V	90000	
26	JPEG	V	90000	
27	unassigned	V		
28	nv	V	90000	
29	unassigned	V		
30	unassigned	V		
31	H261	V	90000	
32	MPV	V	90000	
33	MP2T	V	90000	
34	H263	V	90000	
35-71	unassigned	?		
72-76	reserved	N/A	N/A	
77-95	unassigned	?		
96-127	dynamic	?		
dyn	G726-40	A	8000	1
dyn	G726-32	A	8000	1
dyn	G726-24	A	8000	1
dyn	G726-16	A	8000	1
dyn	G729D	A	8000	1
dyn	G729E	A	8000	1
dyn	GSM-EFR	A	8000	1

dyn	L8	A	var.	var.
dyn	RED	A		
dyn	VDVI	A	var.	1
dyn	H268-1998	V	90000	

APPENDIX B

Session Descriptions

These types of fields contain session specific information. The explanation of these fields and an example has been provided below.

The "v=" field: gives information about the used version of the Session Description Protocol.

v=0

The "o=" field: gives the originator of the session, username and the address of the user's host, a session identifier and version number.

o= jdoe 2890844526 2890842807 IN IP4 10.47.16.5

The "s=" field: shows the textual session name.

s=SDP Seminar

The "i=" field: is optional, it is intended to provide a free-form human-readable description of the session or the purpose of a media stream.

i=A Seminar on the session description protocol

The "u=" field, is optional; if present, it provides a pointer to additional information about the session.

u=http://www.example.com/seminars/sdp.pdf

The "e=" field: is optional; if present, it provides the e-mail address

e=j.doe@example.com (Jane Doe)

The "p=" field: is optional; if present, it provides phone number of the session originator.

p=+1 617 555-6011

The "c=" field: gives the connection information. It can either be added as a single line to SDP or can be given per media level. Format of the c field is:

c=<nettype><addrtype><connection-address>

<nettype> is the pre-defined network type. IN is used for Internet.

<addrtype> is the address type. Thus, SDP can be used for sessions that are not IP based. Current predefined values are IP4 (IP Version4) or IP6 (IP Version 6)

<connection-address> is the address, according to address type.

c=IN IP4 224.2.1.1

The "b=" field: is optional; if present, it provides information about proposed bandwidth to be used by this session or media. The format is:

b=<bwtype>:<bandwidth>

<bwtype> is predefined bandwidth type like AS (Application Specific), CT (Conference Total) etc.

<bandwidth> is the bandwidth in kilobits per second format.

b=AS:66

The "k=" field:is used for conveying encryption keys, if session transported over a secure and trusted channel. This field is not recommended to use. The format is:

k=<method>

k=<method>:<encryption key>

The "z=" field:is used for timezone daylight savings. The format is:

z=<adjustment time><offset><adjustment time><offset>

Example: z= 2882844526 -1h 2898848070 0 which means that at time 2882844526, sessions time will be shifted back by 1 hour, and at 2898848070 original time will be restored.

Time Descriptions

These fields give information about time and repeat period of the session.

The "t=" field:is used for sessions start and stop times. The format is:

t=<start-time><stop-time>

t=3034423619 3042462419

The "r=" field:is used for repeat times of the session. The format is:

r=<repeat interval><active duration><offsets from start-time>

r=7d 1h 0 25h which means that this session will be repeated after 7 day. Active duration will be 1 hour.

Media Descriptions

There can be zero or more media lines in SDP. The format of the “m” line is:

m=<media><port><proto><fmt>

<media> is the media type. Current supported values are audio, video, text, application, and message

<port> is the transport port to that the media stream will be sent

<proto> is the transport protocol like RTP/AVP (used for RTP running over UDP/IP), RTP/SAVP (Secure Real-time Transport Protocol), or UDP (an unspecified protocol over UDP).

<fmt> is a media format description. It changes according to <proto> value. If the <proto> is RTP/AVP or RTP/SAVP the <fmt> contains RTP payload type numbers.

m=audio 48000 RTP/AVP 0 18

m=video 49170 RTP/AVP

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List of Publications and Patents:

- Demirtaş, E. H., 2007: Wireless Energy Systems, B.Sc. thesis, ITU Telecommunication Engineering

PUBLICATIONS/PRESENTATIONS ON THE THESIS

- Demirtaş, E. H., and Oktuğ, S.F, "Providing QoS to Secondary Users Employing VoIP Applications in Cognitive Radio Networks", ICSNC 2011, Barcelona, Spain, October 2011