Studies for the Design of Cognitive Radio Networks for QoS Guaranteed VoIP Communication

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Tamal Chakraborty

This thesis work is dedicated to

My parents Tapas Kumar Chakraborty and Pratima Chakraborty, my sister Tanima Talapatra and brother-in-law Angshuman Talapatra and my niece Antara Talapatra

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List of Publications

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Journals:

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[2] **Tamal Chakraborty** and Iti Saha Misra, "QoS Aware Spectrum Handoff Strategies for VoIP Communication over CR Networks: An Analytical and Test-bed Approach," communicated to **IEEE Transactions on Mobile Computing**, 2016. (SCI Indexed).

[3] **Tamal Chakraborty** and Iti Saha Misra, "A Real-time Spectrum Handoff Algorithm for VoIP based Cognitive Radio Networks: Design and Performance Analysis", IOP Conference series: Material Science and Engineering Journal, vol. 120, 2016. (Thomson Reuters-Web of Science and Elsevier Scopus Indexed).
(**IEEE and URSI Young Scientist Award – 1st Prize**).

[4] **Tamal Chakraborty** and Iti Saha Misra, "Design and Analysis of Channel Reservation Scheme in Cognitive Radio Network," **Computers & Electrical Engineering Journal, Elsevier**, vol. 42, pp. 148-167, February 2015. (SCI Indexed).

[5] **Tamal Chakraborty**, Iti Saha Misra and Salil Kumar Sanyal, "QoS Enhancement Techniques for Efficient VoIP Performance in Cognitive Radio Network," International Journal of Computer Information Systems and Industrial Management Applications, **Machine Intelligence Research (MIR) Labs, USA**, vol. 6, pp. 413-426, 2014. (Elsevier Scopus Indexed).

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[9] **Tamal Chakraborty** and Iti Saha Misra, "Designing a real-time spectrum handoff algorithm for VoIP based Cognitive Radio Networks," Proc. of Radio and Antenna Days of the Indian Ocean (IEEE RADIO), pp.1-2, 2015, **Mauritius**. Also in **IEEE** Xplore: http://ieeexplore.ieee.org/xpl/articleDetails.jsp?arnumber=7323370 (**International Travel Grant Award by SERB, Govt. of India**)

[10] **Tamal Chakraborty** and Iti Saha Misra, "A Priority Based Adaptive Channel Reservation Algorithm for Improved System Capacity in Cognitive Radio Networks," Proc. of IEEE 17th International Conference on Computational Science and Engineering (CSE), pp. 401-406, December 2014, Chengdu, **China**. Also in **IEEE** Xplore: http://ieeexplore.ieee.org/xpl/articleDetails.jsp?arnumber=7023612

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[12] **Tamal Chakraborty** and Iti Saha Misra, "An Analytical Framework for Channel Reservation Scheme in Cognitive Radio Network," Proc. of IEEE 2nd International Conference on Advances in Computing, Communications and Informatics (ICACCI-2013) pp. 127-132, August 2013, Mysore, India. Also in **IEEE** Xplore: http://ieeexplore.ieee.org/xpl/articleDetails.jsp?arnumber=6637158

[13] **Tamal Chakraborty**, Iti Saha Misra and Salil Kumar Sanyal, "Selection of Optimal Transmission Time in Cognitive Radio Network for Efficient VoIP Performance," Proc. of 5th International Conference on Computers and Devices for Communication (IEEE CODEC), pp. 1-4, December 2012, Kolkata, India. Also in **IEEE** Xplore: http://ieeexplore.ieee.org/xpl/articleDetails.jsp?arnumber=6509230

[14] **Tamal Chakraborty**, Iti Saha Misra and Salil Kumar Sanyal, "Design and study of VoIP model in cognitive radio network under different simulation platforms," Proc. of ACM CUBE International Information Technology Conference, pp. 160-165, September 2012, Pune, India. Also in **ACM** Digital Library:http://dl.acm.org/citation.cfm?id=2381748 (**Best Paper Award**).

[15] **Tamal Chakraborty**, "Analysis of Critical Factors for Successful Implementation of VoIP Applications in Cognitive Radio Networks," Proc. of National Conference on Sustainable Development through Innovative Research in Science and Technology, 2012, Kolkata, India.

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List of Patents

(From the contents of the Dissertation)

[1] **Tamal Chakraborty** and Iti Saha Misra, "An Integrated Cognitive Radio System And Method For Quality Of Service Guaranteed Voice Over IP Communication With Enhanced Spectrum Utilization," Indian Patent Application No. 201631003629, Journal No. 10/2016, 4th March, 2016.

[2] **Tamal Chakraborty** and Iti Saha Misra, "Apparatus and Method for VoIP based Two-tier Cognitive Radio Network for Improved Spectrum Utilization" in drafting phase for Indian Patent Application.

List of presentations in International Conferences held abroad

(Related to thesis)

[1] **Tamal Chakraborty** and Iti Saha Misra, "Designing a real-time spectrum handoff algorithm for VoIP based Cognitive Radio Networks," Proc. of Radio and Antenna Days of the Indian Ocean (IEEE RADIO), pp.1-2, 2015, **Mauritius**.
(**IEEE and URSI Young Scientist Award – 1st Prize**).

[2] **Tamal Chakraborty** and Iti Saha Misra, "A Priority Based Adaptive Channel Reservation Algorithm for Improved System Capacity in Cognitive Radio Networks," Proc. of IEEE 17th International Conference on Computational Science and Engineering (CSE), pp. 401-406, December 2014, **Chengdu, China**.

List of presentations in International and National Conferences held in INDIA

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[1] **Tamal Chakraborty** and Iti Saha Misra, "An Analytical Framework for Channel Reservation Scheme in Cognitive Radio Network," Proc. of IEEE 2nd International Conference on Advances in Computing, Communications and Informatics (ICACCI-2013) pp. 127-132, August 2013, **Mysore, India**.

[2] **Tamal Chakraborty**, Iti Saha Misra and Salil Kumar Sanyal, "Selection of Optimal Transmission Time in Cognitive Radio Network for Efficient VoIP Performance," Proc. of 5th International Conference on Computers and Devices for Communication (IEEE CODEC), pp. 1-4, December 2012, **Kolkata, India.**

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[5] **Tamal Chakraborty**, Atri Mukhopadhyay, Suman Bhunia, Iti Saha Misra and Salil Kumar Sanyal, "Analysis and Enhancement of QoS in Cognitive Radio Network for Efficient VoIP Performance," Proc. of World Congress on Information and Communication Technologies (IEEE WICT 2011), pp.904-909, 11-14 December 2011, **Mumbai, India**.

CERTIFICATE FROM THE SUPERVISOR

This is to certify that the thesis entitled, "***Studies for the Design of Cognitive Radio Networks for QoS Guaranteed VoIP Communication***" submitted by **Shri Tamal Chakraborty**, who got his name registered on **9^{th} March, 2012** for the award of Ph.D. (Engg.) degree of Jadavpur University is absolutely based upon his own work under the supervision of **Prof. (Dr.) Iti Saha Misra** and that neither his thesis nor any part of the thesis has been submitted for any degree/diploma or any other academic award anywhere before.

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List of Abbreviations and Acronyms

Abbreviation	***Description***
ACR:	Adaptive Channel Reservation
A/D:	Analog-to-Digital
AM:	Amplitude Modulation
AP	Access Point
BS:	Base Station
CAC:	Call Admission Control
CCC:	Common Control Channel
Codec:	Compressor/de-compressor
CPE:	Consumer Premise Equipment
CR:	Cognitive Radio
CRAHN:	Cognitive Radio Adhoc Network
CRCN:	Cognitive Radio Cognitive Network
CRN:	Cognitive Radio Network
CRN_RESERV:	CRN with the channel reservation policy
CRN_UNRSERV:	CRN without any channel reservation policy
CSA:	Concurrent Spectrum Access
CS-ACELP	Conjugate-Structure Algebraic-Code-Excited Linear Prediction
CSI:	Channel State Information
D-OSA:	Distributed Opportunistic Spectrum Access
DSA:	Dynamic Spectrum Allocation
DSL:	Dynamic Spectrum Leasing
DSP:	Digital Signal Processor
ETSI:	European Telecommunications Standards Institute
FCC:	Federal Communications Commission
FEC:	Forward Error Correction
FM:	Frequency Modulation
FSA:	Fixed Spectrum Access Policy
GSM:	Global System for Mobile Communications
HTML:	Hypertext Markup Language
HTTP:	Hypertext Transfer Protocol
ICT:	Information and Communications Technologies

Abbreviation	*Description*
IEEE:	Institute of Electrical and Electronics Engineers
iLBC	Internet Low Bitrate Codec
IM:	Instant Messaging
IoT:	Internet-of-Things
IP :	Internet Protocol
ISP:	Internet Service Provider
ITSP:	Internet Telephony Service Provider
ITU:	International Telecommunication Union
ITU-T:	International Telecommunication Union - Telecommunication Standardization Sector
KKT:	Karush–Kuhn–Tucker
LAN:	Local Area Network
LTE:	Long Term Evolution
MAC:	Medium Access Control
MLPP:	Multilayer Precedence and Preemption
MMPP:	Markov Modulated Poisson Process
MMS:	Multimedia Messaging Service
MOS:	Mean Opinion Score
MS:	Mobile Station
NEWT:	Network Emulator for Windows Toolkit
NGN:	Next Generation Networks
NRT:	Non Real-Time
OFDM:	Orthogonal Frequency Division Multiplexing
OPNET:	Optimized Network Engineering Tools
OSA:	Opportunistic Spectrum Access
PACR:	Priority based Adaptive Channel Reservation
PBA:	Priority based Allocation
PBX:	Private Branch Exchange
Periodic_msg:	Periodic Message Passing Algorithm
PHY:	Physical layer
PITR:	Primary Interfered Time Ratio
PLC:	Packet Loss Concealment
ProReact:	Proactive and Reactive Handoff

Abbreviation	*Description*
PTT:	Push-To-Talk
PU:	Primary User
Q.O.:	Queue Occupancy
QoE:	Quality of Experience
QoP:	Quality of Perception
QoS:	Quality of Service
QPSK:	Quadrature Phase Shift Keying
RED:	Random Early Detection
RSSI:	Received Signal Strength Indicator
RF:	Radio Frequency
RR:	Receiver Report
RT:	Real-Time
RTCP:	Real-time Transport Control Protocol
RTP:	Real-Time Transport Protocol
SC:	Spectrum Controller
SDR:	Software Defined Radio
SG:	Smart Grids
Simple_msg:	Simple Message Passing Algorithm
SIP:	Session Initiation Protocol
SMS:	Short Message Service
SNR:	Signal-to-noise ratio
SR:	Sender Report
STD:	State Transition Diagram
SU:	Secondary User
TCP/IP:	Transmission Control Protocol / Internet Protocol
TCS:	Target Channel Sequence
TDM:	Time Division Multiplexing
TV Band:	Television Bands
TVWS:	TV White Spaces
UAC:	Underwater Acoustic Communication
UDP:	User Datagram Protocol
UHF:	Ultra High Frequency

Abbreviation	*Description*
VAD:	Voice Activity Detection
VAST:	VoIP based Adaptive Sensing and Transmission
VHF:	Very High Frequency
VoIP:	Voice over Internet Protocol
WAN:	Wide Area Network
WARP:	Wireless Open-Access Research Platform
WBAN:	Wireless Body Area Network
WiMAX:	Worldwide Interoperability for Microwave Access
WRAN:	Wireless Regional Area Networks
WSN:	Wireless Sensor Networks
WWW:	World Wide Web

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Abstract

Wireless communication has witnessed a phenomenal growth in modern era, with the ultimate focus on ensuring seamless connectivity across every single device on the globe. The unique benefits of "going wireless" are increasingly realized among the masses with its distinctive features of portability, cost-effectiveness, mobility and manageability. At the same time, integrated applications in almost every sector of modern life pledge their association with this communication realm. This ensures the widespread popularity of such wireless networks and consequently generates a tremendous potential for continuous research towards capacity scaling of these networks with improved service support. Specifically from the communications perspective, traditional wired and wireless telephony have gradually yielded to the emerging IP (Internet Protocol) telephony, where human voice is transmitted as packets across the Internet following the Voice over Internet Protocol (VoIP). Blessed with infrastructure convergence, the popularity of these VoIP applications has recorded an emphatic growth with its cost-effective features and rapid integration to different types of multimedia and essential services. However, every VoIP session demands strict Quality of Service (QoS) requirements with respect to delay, jitter, packet loss and MOS (Mean Opinion Score), all of which must be maintained within their permissible threshold limits. This has imposed severe restrictions on the traditional wireless networks having limited radio resources, where lending support to these increasing number of VoIP users with QoS constraints becomes nearly impossible. Considering the spectrum resource as a scarce entity, the fixed spectrum allocation policy as followed in such highly complex and dynamic networks, has further aggravated the spectrum scarcity crisis and introduced the spectrum congestion problem. Subsequent studies on the search for potential solutions have finally led to the birth of Cognitive Radio Networks (CRNs) that deploy dynamic spectrum allocation policies to exploit different under-utilized spectrum bands in order to increase the overall spectrum utilization and consequently reduce the issues of spectrum congestion. This makes CRN a

suitable candidate for hosting VoIP communication, where the unlicensed or Secondary Users (SUs) can opportunistically utilize the idle spectrum bands in the absence of the licensed or Primary Users (PUs) and successfully execute VoIP communication without causing harmful interference. Considering the enormous significance of such a system, there is an immediate requirement to explore the possibilities of implementing VoIP over CRN. The focus lies on developing QoS aware spectrum management policies that will coordinate jointly with the application-level parameters towards ensuring QoS guarantees for VoIP sessions by SUs, while protecting the PU traffic from interference at the same time. Therefore, the overall objective of this thesis is to sustain long duration VoIP communication among the SUs with sufficient call quality even under occasional PU presence. However, the primary challenge lies in maintaining this trade-off between real-time SUs and randomly arriving PUs. Thus, VoIP over CRN is truly a challenging yet promising aspect of next generation wireless communication which has recently triggered active interest among the research community and forms the focus of study in this thesis.

As PU arrival leads to unwanted disruptions in VoIP communication, CRN is characterized with the PU detection operation, where each SU deploys its Cognitive Radio (CR) module to evaluate the channel busy/idle characteristics at the onset of every transmission. These alternate modes of sensing and transmission operations are unique to the CR setup and can adversely affect the QoS of the VoIP calls. Hence, the first objective of this thesis is to conduct in-depth studies of VoIP transmissions over a basic CRN scenario. However, this study is hindered by the lack of a comprehensive simulation platform for such VoIP based CR systems in literature. Accordingly, real life-like implementation models are designed and implemented over OPNET Modeler 16.0.A. and Visual C++ platforms, that record a strong correlation between VoIP QoS parameters (delay, jitter, packet loss, throughput, etc.) and CR timing cycle values (sensing duration, transmission intervals). As the SU throughput degrades under the basic CR timing cycle, proactive codec adaptation algorithm is designed to adaptively control the codec bit rates for VoIP SUs. Simulation outcome in the designed model *confirms reduction in throughput degradation after application of this algorithm which upholds the quality of VoIP call in CRN.*

Precisely focusing on the inability of the basic CR timing cycle towards providing adequate QoS for VoIP traffic, the second objective of this thesis is to devise suitable enhancements in the timing parameters. This is fulfilled through the design of a two-phase algorithm. It incorporates momentary sensing slots in the first phase, followed by adaptive variation of sensing and transmission durations using feedback parameters in the second phase. This algorithm is duly accompanied by another algorithm that optimally configures the transmission time and decides its suitability before initiating VoIP call in a particular channel. Subsequent implementation in simulation models leads to the inference that all the fundamental QoS metrics namely, delay, jitter and packet loss are reduced to their threshold limits, thus providing enhanced call quality though at the cost of lower system throughput. This is also confirmed by the designed QoS metric "cog_cap" which takes into account both the VoIP call quality (in terms of R-Factor values) and channel occupancy percentage to measure the "cognitive capacity" of VoIP users in CRN.

Thereafter, viewing the research problem from the systems perspective where CRN comprises of several channels, the next objective is to ensure efficient channel allocation to VoIP SUs so as to minimize the arrival of PUs in the SU occupied channels. Accordingly, the PU based channel reservation policy is studied with respect to VoIP applications based on an analytical framework. Simulation studies record reduced instances of channel switching by SUs (that happens when PU arrives in the SU occupied channel) under the effect of reserving channels. As the reserved channels are inaccessible to the SUs, the drawback of static channel reservation is the loss in system capacity, which is further alleviated through the design and implementation of a novel PACR (Priority based Adaptive Channel Reservation) algorithm. Equipped with ACR (Adaptive Channel Reservation) and PBA (Priority based Allocation) strategies, PACR increases the system utilization by dynamically reserving the channels based on PU traffic activities and thereafter, admitting both Real-Time (RT) VoIP SUs and Non Real-Time (NRT) Data SUs in the CRN.

Further investigations with respect to RT and NRT SUs reveal that the total system capacity in CRN is bounded by several factors. Consequently, the spectrum management policies fail to record significant performance efficiency once the system reaches its capacity limit. Therefore, the next objective of this thesis is to increase the system capacity beyond the established limit by introducing the innovative design concept of "2-tier CRN". This system exploits the silence suppression characteristics of VoIP SUs (denoted by SUtier1) and allows NRT Data SUs (denoted by SUtier2) to utilize the idle frequency bands, when the VoIP SUs are silent. The interaction between these SUs is established through two message passing algorithms, namely Simple_msg and Periodic_msg. Extensive studies using mathematical, Markov and OPNET based simulation models, followed by real test-bed implementation establish the practical significance of the proposed system with increased spectrum utilization compared to earlier works in literature.

So far, this thesis has considered VoIP communication by SU over a single channel under the various effects of PU traffic, channel conditions, VoIP parameters, etc. However, when a PU arrives in such a channel, the SU must vacate the channel and resume the interrupted VoIP call in a new target channel at the earliest. Otherwise, the call is dropped. Thus, spectrum mobility constitutes an integral aspect of CRN design and forms the next focus of work, where the objective is to deploy QoS aware spectrum handoff policy for VoIP users. In this aspect, a real-time integrated spectrum handoff algorithm (to be executed by SU) is designed comprising of three parts, namely i) VAST (dealing with adaptive sensing and transmission followed by three-level dropping decision policy), ii) ProReact (two-phase spectrum handoff scheme comprising of proactive and reactive handoff operations), and iii) Early Call Acceptance policy (for early resumption of VoIP call in the new channel post handoff). This is suitably complimented with joint target channel selection and allocation policies (to be executed by the Spectrum Controller or SC) where the Target Channel Sequence (TCS) is formulated as Fractional Knapsack problem and solved with the proposed greedy approach based GA_TCS algorithm. Extensive studies in analytical and simulation models are performed for the integrated system consisting of SCs and SUs. A drastic reduction in handoff delay and *dropping probability is recorded compared to earlier works in literature, as the proposed algorithms succeed in sustaining VoIP communication even during spectrum handoff.*

Finally, the practical significance and applicability of the proposed research study in this thesis is established through the design of VoIP SU Prototype and PU traffic model, followed by their implementation in a generic CRN test-bed. The SU prototype is equipped with optimally configured parameters (sensing and transmission parameters, RSSI threshold values, handoff parameters, etc.) and formulated design policies (VAST, ProReact, etc.). The designed model efficiently executes VoIP calls with significant improvement in call quality and call acceptance ratios for the SUs under different traffic activities of the PUs. Comparative performance evaluation establishes both the superiority as well as the novelty of the proposed design methodology in this thesis.

Chapter 1.

INTRODUCTION

Chapter Highlights

Motivation

Existing Works

- Capacity Analysis [1.28, 1.29, 1.30, 1.31, 1.34]
- VoIP Traffic Modeling [1.28, 1.29, 1.36, 1.37]
- PU Traffic Arrival Estimation [1.35, 1.38]
- Modeling and Evaluation [1.39]
- Tools used: Mainly Analytical

Required

- QoS Studies and Enhancement
- VoIP parameter optimization (Call signaling protocols, Codecs, Queue Management, etc.)
- Aspects of CRN (cross-layer issues, timing parameters, spectrum handoff)
- Integrated Study of both PU and SU metrics
- Validation in Simulation and Hardware test -bed

CHAPTER 1: Introduction

"The Internet is becoming the town square for the global village of tomorrow."

-Bill Gates, Microsoft

Outline of the Chapter

Since time immemorial, mankind has opened newer frontiers in science and technology leading to innovations that were apparently considered impossible to achieve. This is, perhaps, best understood in a historical context that traces the evolution of early humans from their usage of very simple tools to the modern-day deployment of the complex, large-scale networks that have led to drastic improvement in the living standards. Human behavior, work and society are constantly being upgraded by these technological innovations. Communication is one such realm where revolutionary ideas have been implemented with the sole aim of connecting mankind separated by space and time. And these supporting technologies are called communication technologies [1.1], which are being introduced at an incredible rate and offer increasing efficiency and connectivity for businesses, governments, and individuals. Every time a telephone call is made, a television is watched, or a personal computer is used, the benefits of communication technologies are being received.

At the same time, Connectivity—whether the Internet or mobile phones—is increasingly bringing market information, financial aspects, health and other essential services to remote areas, and is helping to shape people's lives in unprecedented ways. New Information and Communications

Technologies (ICT) [1.2], in particular, the high-speed Internet, are changing the way companies do businesses, transforming public service delivery and democratizing innovation. Market survey reveals that with every 10 percent increase in high speed Internet connections, economic growth increases by 1.3 percent. The mobile platform is also emerging as the single most powerful way to extend these economic opportunities and key services to millions of people. Globally, these mobile devices have transformed from just a means of voice communication to a multi-functional device that allows users to engage in voice/video conferencing sessions, multimedia entertainment applications, financial services, health tracking, news and information sharing, and many more. As per the International Telecommunication Union (ITU) estimates [1.3], the global mobile phone subscriptions reached almost 7 billion in 2014. Overall, the growth in penetration over the past two decades reflects how integrated these devices have become in today's lives. On top of that, traditional telephony methods are quickly yielding to Voice over IP (VoIP) technology [1.4] (also known as IP Telephony or Internet Telephony) where voice is transmitted via packets over the Internet or any IP (Internet Protocol) based networks. It makes international and multi-branch voice communication a lot cheaper and easily maintainable. Studies have shown that, rather than using a Public Switched Telephone Network (PSTN) line, using VoIP can potentially save up to 40% on local calls and up to 90% on international calls [1.4].

However, "every rose has its thorn" and ICT is not an exception. Specifically, the proliferation of the wireless communication technologies has put a serious question mark on the availability of Radio Frequency (RF) spectrum for admission of newer services, thus pointing to the spectrum scarcity crisis as the limiting factor in this regard [1.5, 1. 6]. Spectrum congestion has thus invoked a serious battle for occupancy of the licensed spectrum among business vendors, service providers and individuals. However, a closer look into the problem reveals that spectrum utilization is actually lesser than anticipated, when examined not only from the frequency domain but also the spatial and temporal domains. Consequently, this has generated tremendous interest in the research community to achieve efficient spectrum usage using "Dynamic Spectrum Allocation (DSA)" policies [1.7, 1. 8]. Cognitive Radio Network

(CRN) [1.8] is one such emerging technology that is based on the fundamental principle of DSA and promises to revolutionize the entire communication industry in the near future with the deployment of innovative solutions such as Software Defined Radios (SDR), Cognitive Radios (CR), overlay and underlay spectrum access methods, etc.

It is, therefore, imperative from these discussions that both ICT services (in terms of VoIP) and DSA schemes (in the form of CRN) will play a major role towards shaping the future of human lives. However, both these technologies bring with them additional challenges that must be fulfilled to realize their true potential on a collective basis. For example, VoIP has strict Quality of Service (QoS) requirements [1.9] that ensure its end-user acceptability. At the same time, it is well known that the popularity of ICT is driven by the overall customer satisfaction [1.2]. CRN, on the other hand, can tend to disrupt this communication due to its inherent property of opportunistic (or sometimes concurrent) channel access, leading to degraded call quality. ***Hence, the aim of this thesis is to address these challenges towards building a scalable and feasible CRN for successfully hosting long-duration VoIP communication with sufficient QoS guarantees.***

In accordance with the focus of research in this thesis, this chapter provides a brief introduction to the overall research problem as dealt with in this thesis. Thereafter, the state-of-art research activities in this problem domain are discussed in detail. Based on the limitations of the existing studies in the literature, the motivation behind the proposed research in the thesis is duly established and the objectives are subsequently formulated to address the different aspects of the problem domain. Finally, the significant contributions of this thesis work are highlighted, followed by a general discussion on the overall organization of the thesis.

1.1 Overview of the Research Problem

Technological advancements in the telecommunication domain have made it possible for people across the globe to communicate among themselves. Wireless networks have further alleviated the problem of managing wired connections. Thereafter, the invention of mobile phones has ensured that this connectivity is not hampered even while roaming from one place to another. Although the overall objective of communication (that is, to connect every person on the earth) has not changed over the years, focus has gradually shifted towards ensuring quality, reliability, robustness, flexibility, security and other aspects of communication and networking.

However, necessary infrastructure required to develop architectural models does not come without investment. As a result, communication networks were initially accessible only to the wealthy people apart from political leaders and public service departments. Gradually, emergence of PSTN based telephones [1.10] followed by digitization of analog communication and evolution of wireless telephone technology across different generations made communication services available to the masses. Networking also witnessed a steady growth with the development of Internet and World Wide Web (WWW), which guaranteed transmission of data from one region to another. Subsequently, Voice over IP (VoIP) technology was established with the integration of Internet and communication technologies in order to reduce the costs of communication and also merge data services with voice. A general architecture of the VoIP supported networks is illustrated in Fig. 1.1.

In view of its merits over traditional telephony services, extensive research has been carried out for deploying and maintaining VoIP in practical networks, which has, in turn, lead to a steady rise in the number of VoIP subscribers [1.11]. Widespread popularity in this domain has also triggered further studies on integrating VoIP with PSTN and cellular networks through the design of suitable interfaces and gateways. Establishing VoIP as a commercial entity for the users can be attributed to two significant market strategies. The first policy is adopted by VoIP service providers where they implement VoIP as a service and develop VoIP servers, compatible phones, PBXs (Private Branch Exchanges), gateways, etc [1.12]. Some providers also create hardware and software based modules to aid the research community in understanding the basics of VoIP telephony as well as for further innovations in this domain. The second strategy is followed by various application providers who integrate VoIP with their applications to expand the consumer base, for example, in realistic gaming and social networking applications.

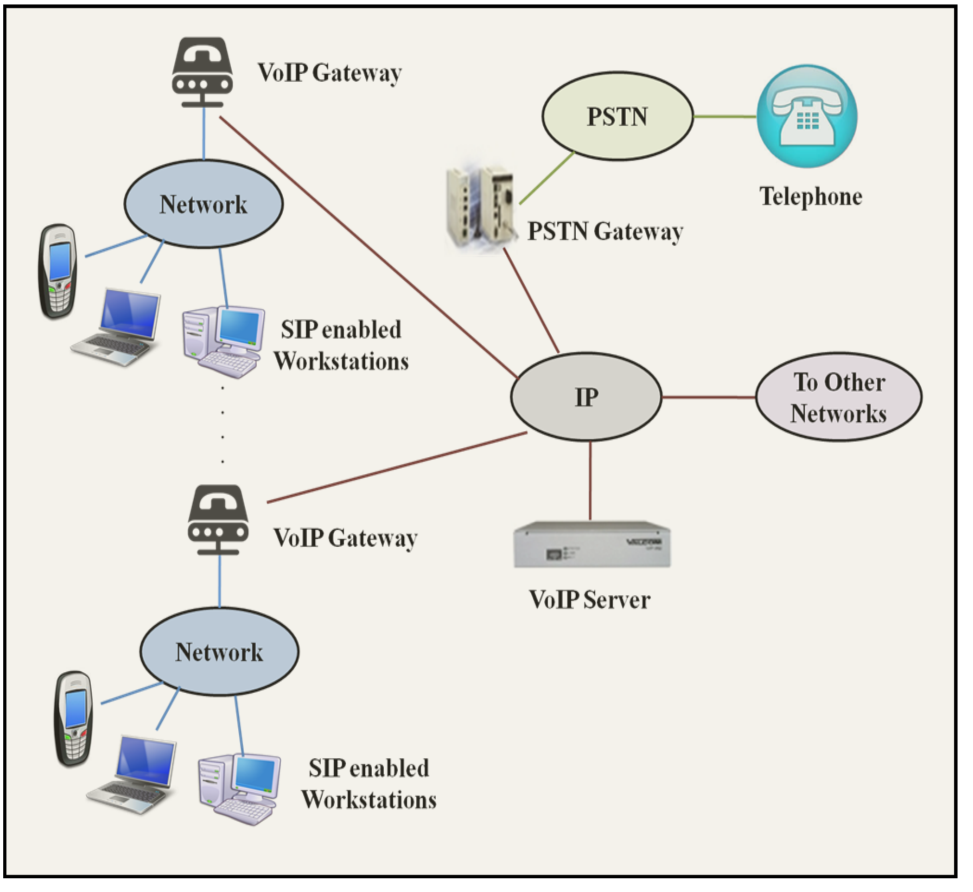

Fig. 1.1 Network Architecture depicting the use of VoIP services

The goals of VoIP implementation are to achieve (a) significant savings in network maintenance and operational costs, and (b) rapid rollout of newer services. Emerging technologies such as Multimedia Messaging Service (MMS), video calling, voice-mail, and different types of VoIP services are currently being taken into use in the markets. Fig.1.2 illustrates these different means of communication. At first, the applications are categorized in three groups, such as calling, messaging, and mailing, based on the nature of the communication. Secondly, the services are differentiated on the type of the content that is conveyed by them, that is, text, image, voice, or video. In the figure, the latency requirements of the services become stricter when moving from bottom to top (from mailing to calling), and the corresponding end-user value rises. On the other hand, capacity requirements increase when moving from right to left (from text to video).

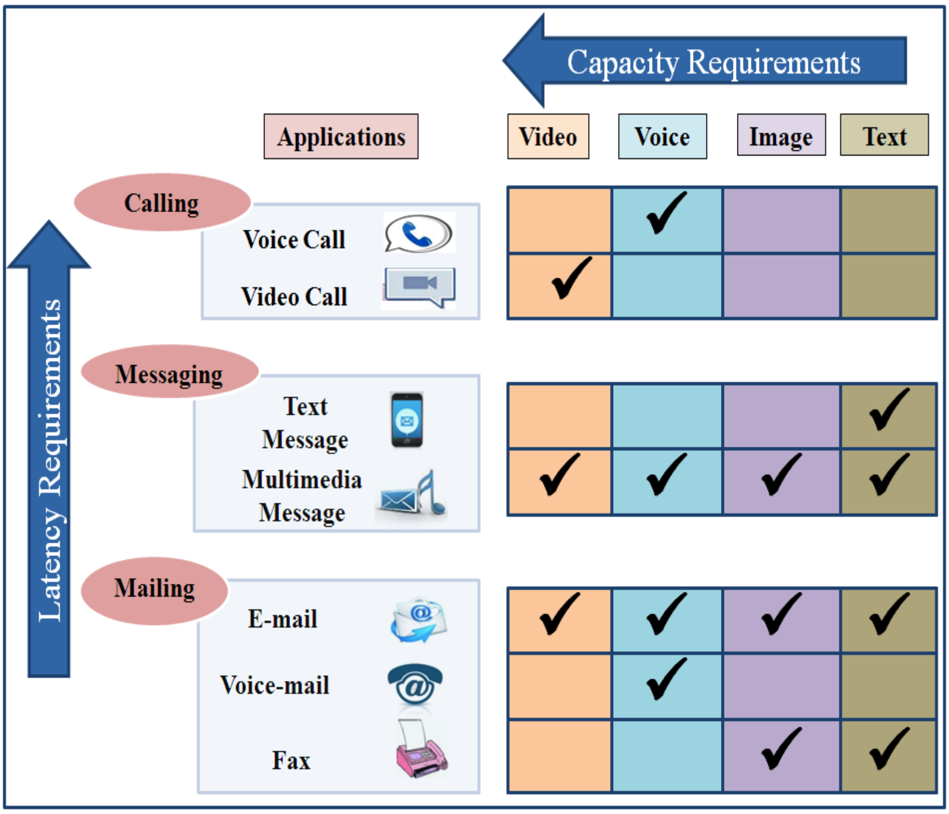

Fig. 1.2 Communication aspects for different applications

A number of different types of VoIP services exist, with various players managing the required network infrastructure and servers. Differences also exist in the pricing schemes, addressing models, level of interconnection to PSTN and mobile networks, and in the regulatory standards. Without going into the specifics of such differences, three fundamentally different classes of VoIP can be recognized in general, named after an analogy to existing, well-known services and systems. The classes are PBX-like VoIP, PSTN-like VoIP, and IM (Instant Messaging)-like VoIP [1.13]. PBX-like VoIP is implemented typically in large enterprises to allow low-cost communication. PSTN-like VoIP replaces the legacy PSTN based telephone services in household and small enterprises whereas IM-like VoIP primarily targets the Internet users. Their characteristics are summarized in Table 1.1.

Table 1.1 VoIP Classes

	PBX-like VoIP	**PSTN-like VoIP**	**IM-like VoIP**
Domain	Fixed, Wireless	Fixed, Wireless	Fixed, Wireless, Mobile
Target users	Large enterprises	Consumers, Small businesses	Consumers
Managed by	Corporation (IP PBX) / local service provider (IP Centrex)	Broadband ISP (Internet Service Provider) / local service provider	Global service provider
Typical pricing scheme	Free calls inside the LAN (Local Area Network). PSTN-like pricing on outgoing calls.	Free / low-cost calls to other VoIP users. PSTN like pricing on outgoing calls.	Free calls to other VoIP users. PSTN-like pricing on outgoing calls.
QoS control	High	Medium / Low	Low
Examples	Cisco CallManager	Vonage, Net2Phone (U.S), Ipon, Sonera	MSN Messenger, Yahoo! Messenger, Skype

However, as with any technological advancement, VoIP suffers from scalability issues that are aggravated by its stringent Quality of Service (QoS) requirements [1.9]. Also, with the advent of emerging networks, VoIP may be implemented over different such platforms that include Wireless LAN, WiMAX (Worldwide Interoperability for Microwave Access), Cellular Networks, LTE (Long Term Evolution) Networks, Cognitive Radio Networks (CRN), 5G networks, etc. Each such system has its own sets of standards and regulations and introduces unique challenges that must be addressed while deploying VoIP over such networks.

Consequently, with a phenomenal increase in the wireless subscribers and swift integration of bandwidth-intensive applications with respect to communication, gaming, social networking and other interactive multimedia services, the problem of spectrum congestion has increased manifold in fixed spectrum assignment policy based wireless networks [1.8]. This problem severely limits the overall system capacity and degrades the transmissions for the ongoing users. Consequently, it has led to various studies on spectrum usage across different frequency bands. As recent Federal Communications Commission (FCC) observations have pointed to large portions of unutilized spectrum in other frequency bands (for example, television broadcasting bands) [1.14], the primary focus in this domain is to solve the spectrum scarcity problem of the traditional wireless networks by providing access to these unutilized frequency slots. This has led to the emergence of Cognitive Radio Network (CRN) [1.8, 1.15] that promises to increase the overall spectrum utilization by allowing opportunistic transmissions of different applications in these vacant spectrum bands. The basic idea of a CRN is to allow unlicensed (or Secondary) users (SUs) to transmit in the available licensed spectrum bands that are allotted to the licensed (or Primary) users (PUs), when the corresponding PUs are absent or idle. The exploiting of spectrum holes by SUs in absence of PUs is highlighted in Fig. 1.3.

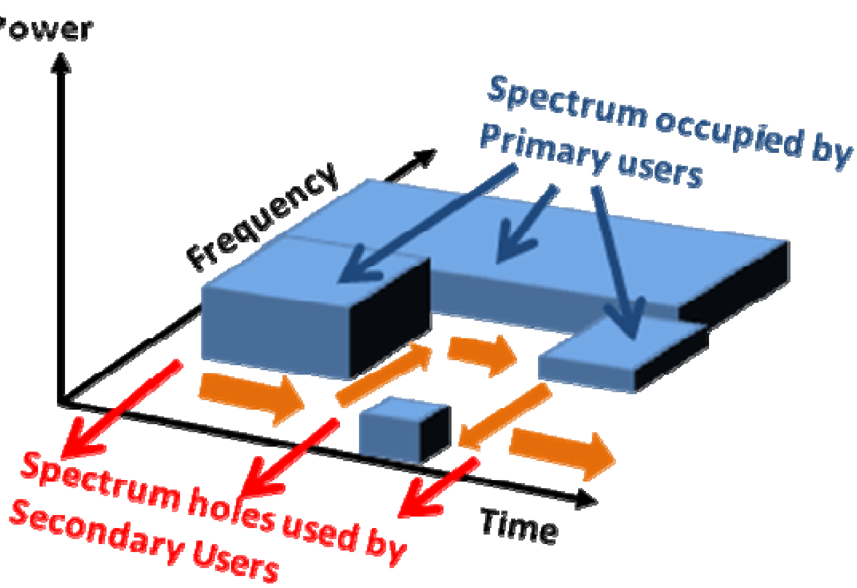

Fig. 1.3 Concept of Spectrum Holes in CRN [1.8]

The first worldwide standard based on the Cognitive Radio (CR) technology is the IEEE 802.22 standard [1.16-1.18] that mainly focuses on the

UHF (Ultra High Frequency)/VHF (Very High Frequency) TV bands between 54 – 862 MHz. This is the licensed frequency spectrum to be used by the SUs on a non-interfering basis, in the absence of PU traffic. The project, formally called the standard for Wireless Regional Area Networks (WRAN) uses the fixed point to multipoint based network topology. Here, a Base Station (BS) serves as the central managing unit and coordinates with all the subscribed users who are termed as the Consumer Premise Equipments (CPEs). In order to offer protection to the licensed PUs, three mechanisms are suggested, that include i) sensing of PU presence, ii) database access and iii) specially designed beacons. Cognitive capability apart from those included in the SUs is also extended to three significant operations namely, i) dynamic and adaptive scheduling of quiet periods, ii) alerting the BS of the presence of PUs by SUs, and iii) the decision by BS to move one or more subscribers out of their current operating channels.

The general architecture of WRAN is shown in Fig. 1.4.

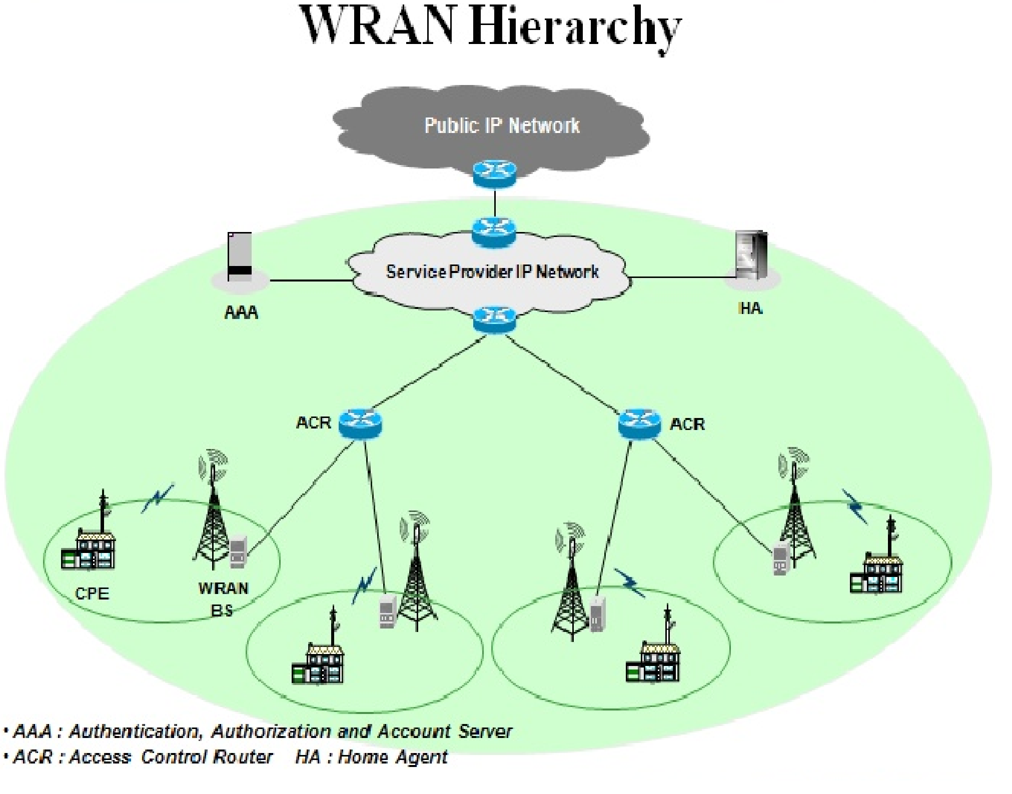

Fig. 1.4 WRAN Architecture [1.17]

The broad domain of CRN however extends to different licensed bands in addition to the TV bands that are already used in IEEE 802.22 standard. A general classification of the CR systems is illustrated in Fig. 1.5.

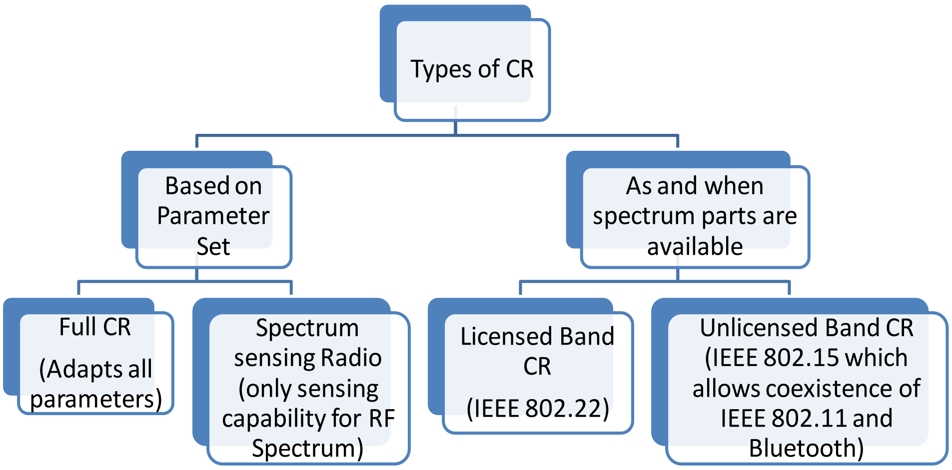

Fig. 1.5 Types of CR systems

Accordingly, spectrum sharing and allocation in CRN can be described under two broad categories. In an underlay based spectrum sharing scheme [1.8], SUs spread their transmission over a wide range of frequency bands while maintaining the interference with PUs below the noise floor. On the other hand, the overlay based spectrum sharing policy [1.8] supports opportunistic usage of idle channels by SUs in the absence of PUs and is considered in this thesis for further evaluation. Taking into account the absolute priority of a PU over a SU with respect to channel access in overlay based CRN, a SU must vacate itself on PU arrival and perform spectrum handoff to shift to another idle channel for its transmission. In absence of a suitable idle channel, this SU is dropped from the network. Thus, arrival of a PU triggers unwanted disruptions in SU transmissions, which may severely degrade the QoS of the underlying applications for SUs. Real-world implementation of CRN, therefore, faces unique challenges in the field of spectrum sensing and analysis, spectrum management, spectrum mobility and sharing [1.8], along with the appropriate tuning of wireless device parameters and regulations on spectrum usage. Notwithstanding these complexities, CRN promises to be an effective platform for hosting VoIP applications, specifically when the number of VoIP users is increasing and the integration of VoIP with other services (gaming, marketing, social networking, business, health) is on the rise.

Thus, it is inferred from the discussion in this section that the integration of VoIP and CRN technologies can prove to be a determining factor towards providing "anytime anywhere" communication to people across the globe, albeit at lesser costs and under lower maintenance overheads. In this regard, this thesis is focused on the "***Studies for the Design of Cognitive Radio Networks for QoS guaranteed VoIP communication***" where the intrinsic design challenges are suitably addressed with respect to maintaining an optimal tradeoff between PU protection and QoS guaranteed VoIP communication by SUs. Specifically, these studies focus on three primary aspects that are highlighted as follows.

- ***Analytical and Simulation based Studies*** of the spectrum management policies with respect to sensing, transmission, analysis, decision and mobility and also their suitability towards providing QoS guaranteed VoIP communication.
- ***Studies involving the Design formulation*** of cross-layer based adaptive strategies involving VoIP and CRN parameters towards enhancing the overall call quality.
- ***Practical Test-bed based Feasibility Studies*** of hosting VoIP applications over CRN without compromising with the call quality for the SUs and without any harmful interference to the PUs.

1.2 State-of-Art of the Research

As already discussed in the previous section, the proposed research study in the thesis provides novel design solutions after conducting extensive studies on the prospects of implementing QoS aware VoIP communication over CRN. Considering the complexities involved in maintaining strict QoS requirements in such opportunistic networks like CRN, the problem domain in this thesis has emerged as a promising yet challenging area of the present day research. Although both CRN and VoIP technologies have emerged as potential winners in their respective fields and attracted widespread research activities, interestingly only limited works have conducted joint studies on such VoIP based CR systems. This section provides a brief overview of these studies and thereby establishes the motivation for the proposed work in this thesis.

It is not without reason that the concept of CR was envisioned in the landmark study by J. Mitola in [1.19] keeping in mind the prospective growth of mobile multimedia communications. In this paper, he has discussed about the advantages of spectrum pooling in solving the spectrum scarcity crisis and providing adequate bandwidth required for different multimedia applications. His work in 1999 invoked another significant study by S. Haykin in [1.20] where the author took over from where Mitola had left his preliminary research on CR. This study has focused on the signal processing aspects of CR and introduced interference temperature as a metric for quantifying and managing the interference in CRN. Also, this paper discusses the aspects of channel-state estimation and modeling, and describes the operations of dynamic spectrum management in coordination with the transmit-power control mechanisms. Both these works subsequently triggered a plethora of contemporary research on CR and CR based systems in relation to spectrum sensing and analysis [1.21-1.23], management [1.24, 1.25] and mobility issues [1.26, 1.27].

However, it was not until 2009-2010 when the benefits of implementing VoIP applications over CRN were truly realized and studied by H.Lee et. al. in [1.28] and [1.29]. In both these works, the reasons for considering VoIP as one of the “candidate applications” in CRN are discussed. This is followed by an extended analysis of the VoIP capacity in CRN where SUs perform VoIP communication in the absence of PUs. The VoIP traffic is modeled as a simple on-off model and matched with the MMPP (Markov Modulated Poisson Process) model using the IDC (Index for Dispersion of Counts) matching techniques in [1.28] and further studied in [1.29] under the conditions of imperfect spectrum sensing by SUs including false-alarms and miss-detections. The wireless channel belonging to the CRN is modeled as a two-state Markov chain where the channel toggles its state from being “Occupied” (by PU) to being “Unoccupied” and vice-versa. Analytical expressions are derived for the average voice traffic generated vis-à-vis the number of wireless channels detected idle by the SUs.

As the vision of executing VoIP calls over CRN finally saw the light of the day, subsequent research studies used this platform to evaluate the capacity from different aspects of the system, for example with respect to retransmissions and multiple-channel effects in [1.30], centralized and distributed channel access schemes in [1.31], MAC (Medium Access Control) protocols in [1.32, 1.33] and scheduling algorithms in [1.34]. Specifically, the user throughput and packet drop probability are analyzed during the calculation of system capacity in [1.30] which allows VoIP traffic to be transmitted across multiple channels. Also, the maximum retransmission limit is set to three to counter the unpredictability in channel availability. This multi-channel concept of VoIP transmission is further studied with respect to different centralized and distributed access schemes in [1.31], where the queuing delay for the voice packets is considered as the primary QoS metric for performance evaluation. Analytical studies confirm an increase in system capacity with multiple voice packet transmissions in a single time slot. Focus is also given to the design of MAC protocols (both contention-based and contention-free) in [1.32, 1.33] so as to minimize the delay during voice sessions in CRN. It is observed that under the effects of independent and correlated channel-state models, system capacity (measured in terms of the number of supported voice users in CRN) increases for the contention-free MAC scheme in comparison to its contention-based counterpart, owing to reductions in collisions between multiple SU transmissions. Shifting from the general trend where there is little or no interaction between PU and SU subsystems, the work in [1.34] uses the concept of SU friendliness, where the primary Base Station (BS) deploys scheduling algorithms so as to shape the spectrum holes in order to host delay-sensitive VoIP traffic by SUs. In this relation, it proposes the incorporation of pricing models applicable to these SUs in order to compensate for the disruptions in PU services.

On a parallel front, another research group has carried out several studies [1.35-1.38] on VoIP transmissions over CRN. In particular, the VoIP traffic is modeled as a simple on-off model and studied under different PU arrival models (on-off and Poisson distribution) in [1.35]. It is proved that the system performance metrics (in terms of packet dropping and blocking probabilities, and packet delays) are further degraded under the Poisson arriving PUs, as compared to on-off based PU traffic, due to higher variance in white space duration under the Poisson model. In addition, using the time-scale decomposition technique and the continuous and discrete-time Markov Chains, an admission control policy with fractional buffering is proposed in [1.36] with the focus on increasing the VoIP Erlang capacity for the admitted SUs in the network. This work is extended in [1.37] where a novel joint packet-level and connection-level model is designed for VoIP traffic. This teletraffic model is developed analytically by talking into account several important parameters in both VoIP (on-off activity processes, periodic packet generation, etc.) and CRN (imperfect spectrum sensing, PU activity detection) domains. Finally, Call Admission Control (CAC) mechanisms for VoIP SUs are devised in [1.38] where it is proved that the inclusion of primary resource occupancy information can lead to significant reduction in call drop probabilities for the SUs.

More recently, two significant works [1.39, 1.40] have been reported in literature, which provide actual implementation of real-time applications over CRN using practical test-beds. Specifically, the RECOG model in [1.39] configures the centrally managed Access Points (APs) with several cognitive functionalities, while bestowing the SUs with two transceivers for simultaneous sensing and transmission, and finally implements VoIP and video streaming applications in the system. On the other hand, the soft real-time model in [1.40] implements only video-streaming applications in CRN under a specific assumption that the SUs are fully aware of the frequency-hopping based PU traffic.

A closer look into the timeline of the literature survey in this section reveals the fact that most of these studies have been conducted in parallel with the proposed research in this thesis. Given the fact that this thesis work commenced from the year 2011 onwards (just one year after the initial work on VoIP based CRN was published by H.Lee et. al.), there was a strong demand for comprehensive studies on QoS aware VoIP communication over CRN, which was unavailable at that point of time. This laid the foundation for the proposed research work in this thesis whose relevance has only grown over these years. Overall, the novelty and significance of this study are established in Fig 1.6 by drawing a comparative analysis of what has already been achieved in literature and which aspects are still lacking and require further investigations for ensuring successful implementation of VoIP services over CRN.

Existing Works

- ❖ Capacity Analysis [1.28, 1.29, 1.30, 1.31, 1.34]
- ❖ VoIP Traffic Modeling [1.28, 1.29, 1.36, 1.37]
- ❖ PU Traffic Arrival Estimation [1.35, 1.38]
- ❖ Modeling and Evaluation [1.39]
- ❖ Tools used: Mainly Analytical

Required

- ❖ QoS Studies and Enhancement
- ❖ VoIP parameter optimization (Call signaling protocols, Codecs, Queue Management, etc.)
- ❖ Aspects of CRN (cross-layer issues, timing parameters, spectrum handoff)
- ❖ Integrated Study of both PU and SU metrics
- ❖ Validation in Simulation and Hardware test - bed

Fig. 1.6 Relevance of the Proposed Research

Based on the outcome of Fig 1.6, the motivation for addressing the research challenges in this thesis is established in the next section.

1.3 Motivation

The primary motivation towards conducting the proposed studies in this thesis is driven by two important factors that include firstly, the deterioration in call quality once and after the traditional wireless networks reach their capacity limits with the ever-increasing number of VoIP users, and secondly, the opportunistic mode of communication as allowed by CRN through dynamic spectrum management policies. In view of the enormous significance of these technologies in the next generation wireless communication as highlighted in Section 1.2, the research community has taken due cognizance of this prospect with several studies being conducted on VoIP based CR systems, primarily with respect to capacity analysis and traffic modeling as briefed in the previous section. However, it is clear from Fig 1.6 that a complete study of VoIP performance in CRN covering all the aspects related to analysis, design and implementation is yet to be carried out and this drives the impetus for the proposed research work in the thesis. Accordingly, this section brings out the key areas of focus in the thesis work by discussing the shortcomings of the relevant works in literature.

To begin with, any analytical study must be suitably verified using real life-like observations either in simulation models or using test-bed prototypes. It is pretty evident from the literature survey of the previous section that the current works have laid more focus on the analytical aspects rather than the implementation details. In general, few simulation models have been developed over CRN using various platforms such as MATLAB and Simulink [1.41, 1.42], OMNeT++ [1.43-1.46], C++ [1.47, 1.48] and NS2 [1.49-1.53] softwares. These models have dealt with several aspects of CRN including the Distributed Opportunistic Spectrum Access in [1.41], CR Mac Protocols in [1.44-1.46], resource allocation with power management and routing protocols in [1.51], etc. ***However, application-oriented simulation studies, especially for real-time VoIP applications in CRN have not been conducted so far. This serves as the initial focal point of this work towards developing comprehensive simulation models for VoIP based CRN that will serve as a dedicated platform for conducting further studies in this discipline.***

Focusing on the VoIP parameters, codecs play a significant role towards shaping the VoIP traffic, eventually affecting the VoIP call quality. Over the years, various codec adaptation algorithms are designed [1.54, 1.55] that reactively switch the codec parameters (for instance, bit rates, switching intervals, etc.) under different conditions such as channel congestion [1.56], affected SU nodes [1.57], priority based sessions [1.58], call admission policies [1.59], etc. ***However, codec switch based on reactive strategy often leads to considerable degradation in user throughput which can be mitigated using proactive approach. Most importantly, none of these works have been executed with respect to the CRN. Therefore, the next focus of research in the thesis is on developing proactive codec adaptation algorithm in order to minimize this throughput degradation and subsequently applying it to VoIP sessions over CRN.***

Next, the CR cycle timing parameters comprising of the sensing interval and transmission duration are considered because suitable configuration of these intervals is integral towards ensuring the success of any CR system. Literature survey related to this problem domain yields several contemporary research studies on analyzing these timing slots based on the sensing-throughput tradeoff problems [1.60, 1.61], calculating optimal duration of data transmissions [1.62, 1.63], outage probability calculations [1.64, 1.65], etc. While throughput is the primary performance metric in most of these works, the effect of delay and jitter for real-time traffic has not been adequately studied. ***Therefore, after developing and analyzing simulation models for VoIP users in CRN, the next step in this thesis is to determine the optimal sensing and transmission durations for protecting the QoS of VoIP SUs, and configure the transmission time before initiating VoIP call in a particular channel.***

Thereafter, the focus shifts from the CR cycle parameters to the studies involving QoS aware spectrum management policies in CRN. PU based channel reservation policy [1.66] is one such strategy that has witnessed limited research activity and forms the basis of this work. In this scheme, a number of idle channels in CRN is reserved for the PUs. On arrival, PUs utilize the reserved channels. Once all the reserved channels are occupied, these PUs access the rest of the idle channels in CRN. Hence, the reserved channels are never allocated to the SUs. The SUs can access only the unreserved idle channels. This ensures interference-free transmission by these SUs for a longer duration of time and reduces the possibility of spectrum handoff, as the probability of arrival of a PU on a channel already occupied by a SU decreases. The major limitation of the existing research works on PU based channel reservation policy [1.66-1.69] is that they have designed Markov Models and applied heuristic approaches to evaluate the system performance, without addressing the practical constraints involved in implementing such a system. Lack of a proper mathematical model further limits extended work in this domain. Also, the efficiency of this mechanism has not been studied from the perspective of real-time applications and finally, the drawbacks of static channel reservation are not addressed in these works. ***This serves as the motivation for the next work in the thesis that focuses on a complete analysis of the PU based channel reservation policy in the CRN with the design of suitable generic mathematical models (after taking into account the practical constraints) and subsequent evaluation of***

the system performance with respect to the real-time VoIP applications. This also leads to the design of dynamic reservation schemes to address the inherent drawbacks in the static policies such as loss in channel utilization, lack of QoS protection, etc.

The ultimate goal of any overlay based CRN system is to maximize spectrum utilization by allowing SUs to access available spectrum when the PUs are absent. Several attempts have been made towards increasing this capacity using call admission control policies [1.70, 1.71], DSA strategies [1.72-1.74] and different power allocation schemes [1.75-1.77]. However, maximum system capacity in CRN is limited due to several factors as derived in [1.78]. It is, thus, implied that the existing policies for the basic CRN fail to record significant increase in performance efficiency once and after the system reaches its maximum capacity limit. This also restricts the system heterogeneity in terms of the different types of users as admitted in CRN from time to time. System heterogeneity in CRN from the user perspective has been addressed in few studies [1.79] that have classified users as high priority (Real Time or RT) and low priority (Non-Real Time or NRT) users and proposed several strategies involving DSA schemes [1.80], priority queues [1.81], virtual queuing interface [1.82] and spectrum allocation framework [1.83]. However, joint studies of system capacity and heterogeneity are yet to be performed specially for VoIP applications in CRN. ***Thus, with the focus on increasing both the system capacity and heterogeneity in terms of total number of users admitted, the next step of this study is to design and implement a novel VoIP based 2-tier CRN by exploiting the silence suppression characteristics of the voice codecs and allowing several SUs to operate concurrently on the same channel.***

Spectrum mobility is another integral aspect of CRN that requires special attention for real-time applications in view of the disruptions caused during channel switch. In this relation, several studies have been conducted with respect to Target Channel Sequence (TCS) calculations [1.84-1.86] and spectrum handoff designs [1.87-1.90]. It is observed that TCS design in these works has not taken into consideration the real-time application requirements such as for VoIP traffic. On the other hand, the spectrum handoff algorithms are based on either proactive [1.87, 1.88] or reactive [1.89, 1.90] policies where each policy has its own set of limitations with respect to call drops, handoff delays and target channel unavailability. Obviously, VoIP transmissions must be guarded against such disruptions induced by spectrum handoff instances and this forms the basis for the next work in this thesis. ***Therefore, motivated by the lack of adequate research on the aspects of the real-time spectrum handoff algorithms for supporting VoIP calls over CRN [1.91, 1. 92], the next aim is to develop an integrated spectrum handoff algorithm combining the advantages of both proactive and reactive policies and supported by QoS aware TCS mechanism, that is specifically tuned for VoIP applications.***

Finally, the last phase of this work deals with the practical aspects of real-life like implementation of VoIP based CR system in the test-bed. It is highly imperative that the actual significance of all the existing works in literature with respect to VoIP based CRN can only be realized through prototype modeling, followed by test-bed implementation. Considering the design complexities in QoS aware policy design and CR modeling, it is obvious that only few works [1.93-1.95] have, indeed, studied CR performance in the test-beds and even fewer [1.39, 1.40] works have implemented real-time applications over such systems. While RECOG [1.39] configures the Access Points with cognitive capabilities to implement VoIP and video-streaming applications, the soft real-time model in [1.40] utilizes the knowledge of the frequency hopping based PU traffic behavior to host video-streaming services. However, the model in [1.39] suffers from the issues of energy efficiency and cost-effectiveness due to the use of two transceivers, whereas the model in [1.40] does not consider a generic CRN and is thus not widely acceptable. In short, none of these works deploy “truly cognitive SU terminals” that must be equipped with sensing, decision, management and mobility functionalities. ***Taking all these factors into account, the focus of this study is on the design of prototype models for SUs and PU traffic, followed by their implementation in a generic CRN test-bed through optimal parameter configuration and policy formulation governed by an enhanced CR principle.***

It is thus inferred from this section that there is a strong motivation for conducting the proposed research studies in the problem domain of VoIP based CRN. In this regard, the primary objectives of this thesis are outlined as follows.

Objectives of the Thesis

- **Development of comprehensive models for VoIP in CRN** under different simulation platforms after exhaustive performance analysis of the parameters involved in CRN and VoIP technologies.
- **Performance evaluation and configuration** of Cognitive Radio cycle timing parameters to ensure high QoS for VoIP applications in CRN.
- **Designing a novel VoIP based "2-tier CRN"** for improved spectrum utilization and analyzing it in Mathematical and Markov models.
- **Evaluation of VoIP parameters** including codecs, call signaling protocols, buffers etc. of the VoIP endpoints and the network elements for performance enhancement over CRN.
- **Design of spectrum management and mobility policies** including channel reservation, target channel selection and handoff schemes for enabling real-time communication in CRN.
- **Prototype Modeling and Practical Deployment of VoIP applications over CRN**, which includes developing different functional blocks after optimal parameter configuration and design policy formulation.

1.4 Significant Contributions of this Thesis

An overview of the comprehensive research work carried out in this thesis towards complete fulfillment of the aforementioned objectives for ensuring QoS guaranteed VoIP communication in CRN is provided in this section.

1. Design and Study of Implementation Models for VoIP based CRN

Methodology:

Simulation Models are developed for VoIP applications in CRN in OPNET

Modeler 16.0.A. [1.96] and Visual C++ following the principles of CRN architecture and subsequently, the critical factors for successful modelling are analyzed. Extensive analysis is performed to study the QoS parameters namely, delay, jitter, packet loss and throughput for VoIP communication in the basic CRN accompanied by a suitable mathematical framework. Proactive codec management is further carried out for minimizing throughput degradation during ongoing VoIP sessions over CRN.

Significant Contributions:

- Analysis shows that there is a strong correlation between VoIP QoS metrics and CR parameters (sensing time, transmission time, etc.)
- Proactive codec bit rate variation increases VoIP communication duration and reduces call drop.
- The developed simulation models can be used for analysis of the existing algorithms along with experimentation of new algorithms with appropriate modifications.
- The designed models can be implemented as training tools to acquaint researchers with the fundamental ideas of hosting VoIP calls in CRN.

Publications:

1. Journals (**SCI Indexed**) **:** Journal of Networks'12, IJCA Journal' 13

2. International Conference Proc. : ACM CUBE'12 (Best Paper Award)

2. Analysis and Enhancement of CR Cycle Timing Parameters

Methodology:

The CR timing parameters (sensing and transmission cycles) are configured through the proposed algorithm comprising of two parts. The first part incorporates momentary sensing slots and the second part adaptively varies the timing intervals based on feedback based parameters to improve call quality while reducing interference with PU traffic. In addition, an algorithm is proposed that addresses the problem of miss-detection during spectrum sensing and performs PU traffic prediction to configure optimal transmission duration for initiating VoIP calls in a particular channel.

Significant Contributions:

- Performance analysis reflects deterioration in call quality for the basic CR timing cycle.
- Delay and jitter are reduced to below 150 ms and 100 ms respectively, after application of the momentary sensing slot in the first part of the proposed algorithm.
- Adaptive strategy in the second part of the algorithm further drops the delay to less than 50 ms, thereby increasing the quality of VoIP call.
- Optimal transmission time detection algorithm reduces both the interference with PU traffic and the number of handoff instances along with reduction in the associated delay.

Publications:

1. Journals : IJCISIM'14 – MIR Labs (Elsevier Scopus Indexed)

2. International Conference Proc. : IEEE WICT'11, IEEE CODEC'12

3. Channel Reservation Scheme for VoIP Users in CRN

Methodology:

A complete study of channel reservation scheme in CRN is performed and the policy is applied for improving the QoS of the VoIP applications. Based on the underlying architectural framework, the advantages of this policy are subsequently applied to design *Priority Based Adaptive Channel Reservation* (*PACR*) algorithm that increases both channel utilization and system heterogeneity. Detailed analysis is performed over analytical and simulation models for establishing the performance superiority with respect to the reported works in literature.

Significant Contributions:

- The number of dropping instances for SUs decreases by 81.88 %.

- Number of erroneous packets received by the SUs decreases by 41.81%.
- Rigorous stochastic analysis of *PACR* Algorithm followed by simulation studies record over 100% improvement in SU Sum Goodput compared to Basic CRN with effective allocation of idle channels to RT VoIP users and NRT Data users.

Publications:

1. Journals (SCI Indexed): Elsevier CAEE' 15

2. International Conference Proc. : IEEE ICACCI'13, IEEE CSE'14

4. Design and Implementation of VoIP based 2-tier CRN

Methodology:

The problem of limited user capacity in basic CRN is addressed by proposing the novel design concept of "VoIP based 2-tier CRN" that allows another tier of SUs to transmit, in addition to the existing set of PUs and SUs. Mathematical and Markov models derive critical system metrics and evaluate their performance in terms of dropping and blocking probabilities. Two message-passing techniques are proposed, namely *Simple_msg* and *Periodic_msg* algorithms to map the complicated interactions among the two tiers of SUs. The overall concept is successfully implemented in a test-bed based disaster management system that establishes its practical significance.

Significant Contributions:

- Mathematical Model: Minimum 100% increase in System capacity for single transmission by SUtier2 in one idle time slot.
- Markov Model: Probability of successful transmission increases for a single SU by 20%.
- Simulation Model: System capacity in terms of throughput (time average) increases for multiple transmissions by SUtier2 in a single idle time slot of SUtier1 under identical traffic rates of all SUs by 262%.
- Test-bed Model: System capacity increases by 494% for different traffic

rates of SUtier1 and SUtier2.

Publications:

1. Patent: "Apparatus and Method for VoIP based Two-tier Cognitive Radio Network for Improved Spectrum Utilization" in drafting stage

2. Journals (SCI Indexed): IEEE SYSTEMS JOURNAL'16, IJACSA'13

3. International Conference Proc. : IEEE COMNETSAT'13

5. Real-time Spectrum Handoff with suitable Channel Selection Strategies

Methodology:

The challenges in spectrum mobility with respect to real-time spectrum handoff for VoIP users in CRN are effectively addressed with the design of joint target channel selection and QoS aware handoff strategies for SUs in CRN. While the Spectrum Controller (SC) node calculates and allocates target channels in real-time using *GA_TCS* algorithm, the SUs communicate with each other and the SC nodes while executing adaptive sensing, transmission, dropping and handoff operations towards maintaining sufficient QoS for ongoing communication. The spectrum handoff algorithm comprises of three parts, namely, i) *VAST* with *three-level dropping decision* policy, ii) *ProReact*: Two-phase spectrum handoff policy, and iii) *Early Call Acceptance* Policy.

Significant Contributions:

- Minimum 10% reduction in call dropping probability.
- Handoff delay remains below 150 ms during the average-case *proactive* phase and 600 ms in the worst-case *reactive* phase (40% and 60% lower than previous works respectively).
- Highest number of target channels selected within threshold limit in optimal *O (n log n)* time by the *GA_TCS* algorithm.
- Ensures minimum 180 ms of effective VoIP communication duration for

every CR cycle of 250 ms.

Publications:

1. **Journals (SCI Indexed) :** IOP Journal'16, Communicated to IEEE Transactions on Mobile Computing' 16.

2. **International Conference Proc. :** IEEE RADIO' 15 (IEEE Young Scientist Award – 1st Prize).

6. Prototype Test-bed Model for deploying VoIP applications over CRN

Methodology:

Finally, the credibility of the overall work is established through robust modeling of SUs and implementing them in a practical hardware test-bed. Based on an enhanced CR principle, the designed policies in CR and VoIP domain are executed in different functional blocks towards developing QoS aware Prototype models for VoIP users and PU traffic models. Performance superiority and novelty of this configured prototype are established in a complete test-bed setup involving PU transmissions, SC nodes and VoIP communication by SUs.

Significant Contributions:

- Threshold limits in Sensing time (max.) = 3 ms, Tx. time (min.) = 40 ms.
- Multiple Sensing time intervals (0.3ms, 3ms), Optimum Threshold values for RSSI = -83.15 dBm, PU Detection Energy=0.00044pJ, 0.0044pJ.
- Suitable multiple Transmission durations (41ms, 410 ms).
- Initial sensing time t_{s_i} = 0.15 ms, Detection Energy=0.00029 pJ.
- Mean QoS metrics such as MOS (= 3), Loss (PU=0.005, SU= 5%) indicate good quality VoIP calls after implementing the proposed algorithms.
- Call Acceptance Ratio Improvement for the SUs = 150%.

Publications:

1. **Patent:** "An Integrated Cognitive Radio System And Method For Quality

Of Service Guaranteed Voice Over IP Communication With Enhanced Spectrum Utilization" filed for Patent.

2. Journal: Indian Patent Journal'16.

1.5 Thesis Organization

The thesis is henceforth organized as follows.

- After the detailed discussions pertaining to the general introduction, motivation, contribution and outline of the thesis in Chapter 1, ***Chapter 2 provides the details related to the background study involving VoIP and CRN technologies and also establishes their relevance in the context of current and next-generation wireless networks***. The evolution of these technologies is traced from the early days of wired telephony to the modern-day communication involving integrated applications and services that have led to the rising problem of spectrum congestion. This is followed by discussions on the technical aspects related to VoIP and CRN that is accompanied by an analytical framework involving the system parameters. In addition, this chapter describes the primary QoS metrics related to VoIP communication and critical system parameters in the CRN domain that will be used for performance evaluation in the subsequent chapters. Finally, the key application areas in relation to VoIP based CR systems are pointed out while also addressing the design challenges in these areas.

- It is evident from the background study of Chapter 2 that there is an enormous scope of research for hosting QoS aware VoIP communication over CRN, thus laying the foundation for the proposed research work in this thesis. Accordingly, the first step towards conducting state-of-art research studies in this domain is the design and development of a real life-like simulation model that will serve as a platform for validating the analytical observations and verifying the performance superiority of the proposed algorithms. Therefore, ***Chapter 3 deals with the design of***

simulation models for hosting VoIP services over CRN. As the literature survey points to lack of comprehensive simulation models for implementing VoIP applications over CRN, simulation models in single-channel and multiple-channel scenarios are developed using distributed and centralized architecture in OPNET Modeler 16.0.A and Visual C++ respectively, duly accompanied with the mathematical framework. Extensive analysis of the VoIP performance over CRN is carried out in terms of the QoS metrics under several conditions including the basic CR timing cycle, SU traffic distribution patterns, different PU activity models, PU detection issues and different packet processing delays with respect to the SU node. Thereafter, a proactive codec adaptation algorithm is proposed involving VoIP codecs and Random Early Detection (RED) [1.97] buffers. Finally, the advantages of the proposed models along with the areas for further improvement are discussed in detail.

- Performance studies in the simulation models of Chapter 3 reflect the limitation of the basic CR timing cycle in providing the necessary QoS required for successful VoIP communication. ***This problem is addressed in Chapter 4 that performs optimal configuration of the sensing and transmission durations in order to maintain the tradeoff between providing QoS guarantees to VoIP SUs and protection to the PU traffic from harmful interference.*** Comparative performance analysis is carried out in the designed simulation models for the enhanced CR timing cycle that is designed in two parts. This is followed by the selection of the optimal transmission duration before initiating VoIP call in a particular channel. Detailed analysis using analytical framework highlights the advantages of the proposed algorithms, while deriving the critical system conditions that must be satisfied for QoS guarantees. This is followed by the design and application of a derived QoS metric "*cog_cap*" that takes into account both the call quality metric (in terms of R-Factor) and the channel occupancy percentage in order to measure the performance superiority of the proposed CR timing cycle algorithm.

- Focus shifts from the CR cycle configuration in Chapter 4 to the application of spectrum sharing and management policies towards supporting QoS aware real-time transmissions in Chapter 5. ***Specifically, Chapter 5 deals with the channel reservation policies in order to ensure maximum throughput for these SUs, while minimizing their interference with PUs.*** After highlighting the pros and cons of PU based channel reservation policy through the design of system model, architectural framework and analytical formulations, the practical applicability of these policies towards providing QoS guarantees to the VoIP users is ensured through extensive studies in simulation and analytical models. In addition, the drawbacks of the static channel reservation policy are mitigated through the design and analysis of a novel *Priority based Adaptive Channel Reservation* (*PACR*) algorithm that is developed in two phases. Finally, detailed simulation studies validate the mathematical outcome and confirm the feasibility and practical applicability of channel reservation schemes towards deploying VoIP services in CRN.

- From the applications perspective, securing adequate QoS for successful VoIP communication by the SUs is the primary target and is fulfilled by the previous chapters. On the other hand, from the systems perspective, increasing the overall spectrum utilization is one of the crucial aspects of CRN that is not yet explored. Due to an upper bound on the maximum system capacity of CRN as observed from the literature survey, there has to be a trade-off between the total number of admissible users in CRN and the level of QoS to be provided to their applications. ***In this relation, Chapter 6 proposes the novel design concept of "VoIP based 2-tier CRN" that further aims to maximize the system capacity over and above the existing limit, by exploiting the silence suppression characteristics of VoIP transmissions.*** The proposed system model is validated using analytical, Markov and OPNET based simulation models, all of which record significant improvement in system capacity without compromising with the VoIP call quality or PU protection issues. Coordination among the SUs belonging to the two tiers is maintained using two novel message-passing techniques. Finally, the practical utility of this system is established with respect to disaster management system by developing a prototype model for the same and implementing it in the hardware test-bed with absolute success.

- After suitably dealing with the CR cycle parameters, channel reservation schemes and capacity enhancement using VoIP parameters, the next important objective is to study the additional constraints imposed by spectrum mobility in CRN, a unique aspect that has already been highlighted in the background study of Chapter 2. This is more so for real-time applications such as VoIP where unwanted disruptions caused due to spectrum handoff can lead to severe degradation in QoS and even call drops under the worst cases. ***In this aspect, Chapter 7 deals with the impact of spectrum mobility towards QoS sensitive VoIP traffic over CRN, and suitably designs a real-time spectrum handoff algorithm (in three parts) jointly with target channel selection and allocation algorithms, that provide QoS support to the underlying VoIP applications in CRN.*** Feasibility studies are conducted for the design methodologies in analytical models that lead to the derivation of important system metrics. Finally, comparative performance evaluation and superiority of the proposed algorithms are studied in simulation models that not only validate the proposed design but also confirm drastic performance improvement over existing works in the literature.

- Previous chapters have already presented novel design techniques that have successfully enhanced the QoS of ongoing VoIP calls in CRN. Also, extensive studies in analytical and simulation models have confirmed their performance superiority with respect to earlier works in literature. However, as with any network design, practical deployment of VoIP based CRN is necessary to conduct the feasibility studies of these algorithms. ***Chapter 8 addresses this issue of practical deployment through the test-bed design and implementation of a generic CRN test-bed where VoIP applications are initiated, managed and terminated successfully by the SUs without interfering with the PU transmissions.***

Both the PU and SU prototypes are modeled based on a modified CR cycle and evaluated in the test-bed after suitable parameter configuration and design policy formulation, that record drastic improvement in VoIP performance over CRN as compared to earlier works in the literature.

- ***Finally, Chapter 9 gives the conclusive remarks on the findings from all the chapters of this thesis dissertation and summarizes the practical significance and applicability of hosting VoIP applications over CRN in the context of next-generation wireless communication.*** The contributions of each chapter closely follow the overall objective of this thesis work. Accordingly, all the significant aspects pertaining to CRN and VoIP have been addressed in these chapters. Also, the content flow of this thesis shows the way in which the VoIP applications can be methodically deployed in CRN. Thus, the ability of CRN to sustain long duration VoIP communication even in the presence of occasional PU activity is established in the thesis and this paves the way for further scope of research in this discipline. Consequently, this chapter finally discusses some of the prospective research opportunities that can be extended over the proposed research work in this thesis.

It is worth mentioning that although an extensive literature survey has been conducted in Chapter 1 in order to establish the motivation behind this work, every subsequent chapter in the thesis performs a more specific literature survey that exclusively deals with the problem under focus in that chapter. This is done to clearly bring out the pros and cons of existing studies, that in turn provide the impetus for carrying out further investigations in that area. Therefore, every chapter is accompanied with a separate "References" section that ensures better readability of the thesis and acquaints the reader with contemporary works in related disciplines.

1.6 Flow of Thesis

The flow of research contributions in this thesis is illustrated by the schematic diagram in the next page.

Research Contributions on VoIP based CRN

Outcome

Implementation in Simulation Models:

Simulation models are designed in OPNET Modeler 16.0.A. and Visual C++, that are used for QoS evaluation of VoIP calls over CRN, followed by RED buffer based Proactive Codec Configuration.

- Strong correlation between VoIP QoS metrics and CR timing parameters
- Basic CR Timing Cycle deteriorates VoIP call quality
- Proactive Codec Configuration reduces throughput degradation during codec switch

Modified CR Cycle Timing Parameters:

At first, proposed algorithm is designed in two parts for momentary sensing and adaptive variation of CR cycle timing parameters. Secondly, optimal transmission time is determined for initiating VoIP calls. Finally, a derived QoS metric, *cog_cap* is designed to evaluate VoIP performance over CRN.

- Delay and Jitter remain within 150 ms and 100 ms respectively.
- Optimal transmission time reduces interference with PU traffic.
- Number of handoff instances is reduced.

PU based Channel Reservation Scheme:

Channel Reservation is applied to protect VoIP SUs. *PACR* Algorithm is designed further to make channel reservation adaptive to PU activity.

- Dropping instances are reduced by 81.88 %.
- Number of Erroneous packets received decreases by 41.81 %.
- *PACR* increases system capacity and protects VoIP QoS.

VoIP based 2-Tier CRN:

Novel design of VoIP based 2-Tier CRN in analytical, simulation and Markov models record increased system capacity. Test-bed model validates its practical applicability.

- System capacity increases by 262% and 464% under identical and different characteristics of SUs.
- Successful Transmission Probability increases by 20%.
- Successful application to disaster management systems.

Real-time Spectrum Handoff:

A three-part spectrum handoff algorithm is designed to minimize QoS degradation during handoff, along with joint target channel selection and allocation algorithms for efficient channel assignments.

- Handoff delay remains below 150 ms.
- Atleast 10% reduction in call drop probability during spectrum handoff.
- Target channel calculation in optimal $O(n \log n)$ time.

Prototype Model Implementation:

Working prototypes for SUs and PU traffic are modeled and implemented in a generic CRN test-bed that establishes the feasibility of the proposed study in this thesis.

- Threshold limits derived for sensing time (0.3ms-3 ms), transmission time (41 ms – 410 ms), RSSI values, handoff response time, etc.
- 55% improvement in MOS, 73% drop in PU packet loss ratio.
- 150% rise in SU call acceptance ratio.

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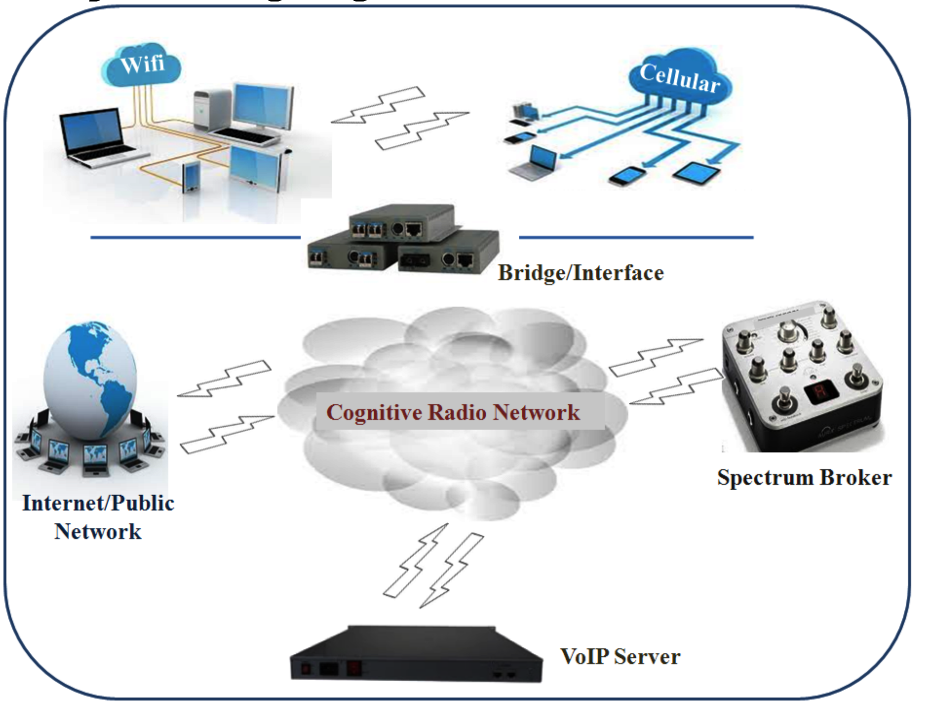
Chapter 2.

BACKGROUND STUDY: VOIP AND CRN TECHNOLOGIES

Chapter Highlights



CHAPTER 2: Background Study: VoIP and CRN Technologies

"The human voice is the organ of the soul."
-Henry Wadsworth Longfellow

Outline of the Chapter

Comprehensive literature survey in the previous chapter has already laid the foundation for the proposed research study in this thesis. It is quite clear from the growing interest in this domain that CRN is envisioned by the wireless community towards providing maximum communication opportunities and is thus poised to become a significant entity in the next generation wireless networks such as 5G networks [2.1]. Thus, the focus of work in this thesis is to deploy the most popular applications like VoIP over this CRN and evaluate the feasibility and practical applicability of such systems. As the success of real-time VoIP transmissions is determined by the overall call quality, the objective is to carry out in-depth studies of QoS metrics (delay, jitter, packet loss, MOS, etc.) jointly with CRN parameters (sensing parameters, transmission parameters, dropping and handoff decisions, etc.) through design and implementation of novel policies concerning both VoIP and CRN technologies.

Accordingly, this chapter provides a technical overview of both these disciplines with special emphasis on their practical relevance and highlights the challenges and complexities involved in such systems. The mathematical background is also established with the modeling of VoIP traffic and wireless channel in CRN. In this relation, the important parameters are discussed pertaining to VoIP QoS and CRN that are subsequently taken into account during the proposed studies in the ensuing chapters.

2.1 Introduction

Rapid advancements in the wireless communication sector have led to the advent of emerging smart networks that focus on building an integrated platform for hosting a wide range of applications. With feature-richness and user-friendliness as their primary criteria, these bandwidth savvy applications continue to attract increasing number of subscribers, thereby leading to high data consumption and contributing to the problem of spectrum congestion [2.2, 2.3]. This problem severely limits the total system capacity while degrading the application-level QoS, thus posing a serious question to the reliability of such systems. CRN [2.4-2.6] promises to solve this spectrum scarcity problem by allowing these applications to access the idle frequency slots in different spectrum bands [2.7], and in doing so, increases the overall spectrum utilization.

At the same time, infrastructure convergence has contributed to the widespread popularity and deployment of all-IP networks as a cost-effective means of implementing IP based mobile communication and networking. VoIP technology (where voice is transmitted as IP packets) [2.8] has thus witnessed utmost significance owing to its low operational and maintenance costs and rapid integration with user-centric applications. Increasing demand for VoIP services makes it a suitable candidate for CRN technology [2.9] and is, therefore, the primary focus of research in this thesis. An ongoing VoIP communication by SU can be disrupted by the untimely presence of PU in the licensed channel. This implies that the SU must halt VoIP transmission, vacate the current channel and perform spectrum handoff at the earliest to a suitable idle channel to resume communication. All these operations must be executed in tandem to avoid QoS degradation and subsequent call drop. ***Therefore, the***

feasibility study and practical applicability of a real VoIP based CR system require thorough examination of critical factors and subsequent formulation of design methodologies pertaining to both VoIP and CRN technologies, thus laying the foundation for the work in this thesis.

In this relation, it is very important to understand why these technologies came into existence in the very first place. Thus, this section provides a timeline based overview of their evolution and thereafter, discusses their relevance with respect to modern-day wireless communications.

2.1.1 Evolution of VoIP

There are two fundamental technologies that led to the discovery of VoIP, namely the telephone and the Internet. Telephony has its origin with telegraphy in 1844, when Samuel Morse developed the capability to send pulses of electric current over wires that spanned distances farther than one could shout, walk, or ride. Voice communication subsequently became possible with the invention of telephone by Graham Bell on March 10, 1876. By 1906, American inventor, Lee De Forest, invented a three-element vacuum tube that revolutionized the entire field of electronics by allowing amplification of signals with respect to both telegraphy and voice. This was followed by the breakthrough execution of wireless voice communication using Amplitude Modulation (AM) during 1920s. The ensuing years saw a tremendous growth in radio station broadcasting that brought the possibility of real-time information to the public. Of course, wires still had their place because radio was not always the most reliable medium due to environmental factors.

Telephone technology progressed steadily, and telegraphy still found a place in data communications in the form of the telegram. Radio technology advanced through the 1930s with the notable invention of Frequency Modulation (FM), which provided better sound quality and was more resistant to interference than the older AM broadcasting system. Thereafter, post World War II period saw an explosion of innovations with the development of the transistor (December 1947) and the birth of the computer. Computers provided a tool for people to process and transfer lots of data at high speed. The Space Age also began with the launch of the Soviet satellite named 'Sputnik' on

October 4, 1957. Satellite communications [2.10] provided reliable long distance communications by augmenting or replacing cables. This generated the interest as well as the demand for reliable, anytime, anywhere communications across the globe. However, it took almost 26 years after the Sputnik event before cellular communications brought mobile voice communications to the masses.

In the early days of telephony, whenever a user wanted to talk to another person, they would ring the operator and give the name or number of the other party. Next, the operator would connect a patch cord (2 wire cable with a jack plug on each end) between the two phones and the two people could communicate. Bundles of wires called trunks ran between exchanges, forming proto-networks. Networks were connected together hierarchically until they connected countries across the world. This was the beginning of the Public Switched Telephone Network (PSTN) [2.11] which is now the worldwide collection of interconnected public telephone networks designed primarily for voice traffic. It is a circuit-switched network where a dedicated circuit is established for the duration of a transmission, such as a telephone call. Originally only an analog system, the PSTN is now almost entirely digital and employs efficient algorithms to ensure the reliability of voice calls.

Parallel growth in networking ensured development of Internet in 1968 by ARPANET, which was followed by the design of HTTP (Hypertext Transfer Protocol) and HTML (Hypertext Markup Language). This marked the advent of the World Wide Web (WWW), which increased the popularity of Internet. The Transmission Control Protocol / Internet Protocol (TCP/IP) [2.12] was created in the year 1971 by Dr. Vint Cerf, that defined the nature of data packets to be sent via Internet and established the rules for routing of packets to their destinations.

With the rapid growth of Internet and deregulation of the telecommunications industry, infrastructure convergence in the form of building voice applications on top of data networks gained momentum. This is suitably illustrated in Fig 2.1. Eventually, this led to the birth of VoIP which allowed voice to be carried by IP packets over any IP based networks such as Internet.

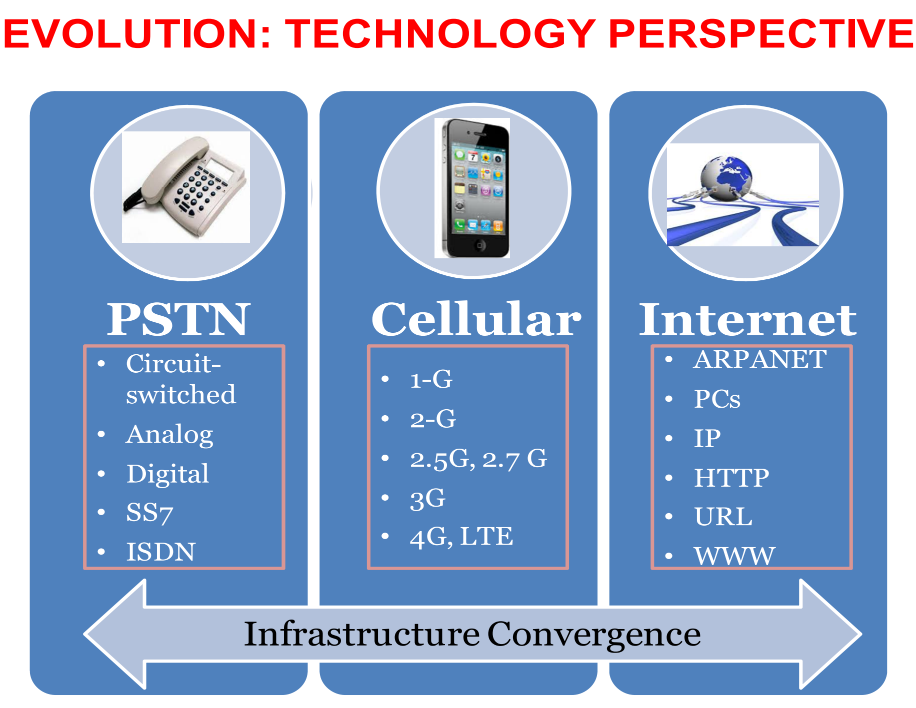


Fig. 2.1 Infrastructure Convergence

2.1.2 Evolution of Cognitive Radio Networks (CRN)

In order to understand the emergence of CRN, it is necessary to detect the problems faced by the service providers in providing services to their registered users in today's world. With rapid integration of various bandwidth intensive applications, the biggest constraint faced by these providers is to render adequate network resources for each such application while still maintaining the promised data rates. For example, the commonly used smartphone may need to compete with several other traffic sources for getting access to the medium during data transmissions in connection with the gaming, video streaming or social networking applications. Because these data transmissions suffer from disruptions, the end-user satisfaction gets severely hampered, which is not desirable.

As "going wireless" is the new buzzword, it must be noted that the wireless spectrum is both scarce and expensive [2.13, 2.14]. In this era of "Internet-of-Things (IoT)", there is an upscale surge in the use of device-to-device communications and vehicular communications that in turn introduces the problem of spectrum congestion. This also limits the total number of admissible users that can be supported by the network.

The realization that the spectrum is finite and limited can easily be visualized by using the “cell” concept. The users that are transmitting in the cell must efficiently share the RF spectrum so that the throughput is maximized, while interference is reduced. However, the problem with wireless transmission is that the spectrum can only be reused to the extent of having smaller cells. Even with such small cells, the spectrum has to be allocated among a significant number of applications with diverse requirements, all of whom belong to the same coverage area. This becomes a highly challenging issue.

Fig. 2.2 illustrates the spectrum allocation map as per the FCC regulations [2.15].

Fig. 2.2 Spectrum Allocation Map by FCC [2.13]

The unique challenges imposed by the RF spectrum can be summarized briefly as follows.

i) Susceptible to overlapping interference that restricts its usage.

ii) Propagation characteristics of different radio waves.

iii) Suitability of different frequencies for different applications.

Thus, one needs to carefully plan the network and draw a limit among the number of supported users and the level of QoS offered to them, while avoiding the problem of spectrum congestion.

The situation becomes even worse from the perspective of the developing nations such as India [2.16]. Here, the availability of spectrum is always a problem, especially when the licensed spectrum is being accessed by an ever-increasing population (that has triggered almost an exponential growth in the use of smartphones and other wireless devices) [2.17]. It is clearly evident from Table 2.1 that the licensed spectrum availability is much lower in India compared to U.S. and European Countries.

Table 2.1 Total licensed spectrum in various countries (in MHz) [2.18]

Country	Current	Pipeline	Hz/Subscriber
USA	608	55	2.1
Australia	478	230	22.8
Brazil	554	0	2.0
China	227	360	0.5
France	555	50	9.3
Germany	615	0	6.2
Italy	540	20	5.9
Japan	500	10	3.3
Spain	540	60	11.8
U.K.	353	265	7.9
India	221	10 (estimate)	0.2

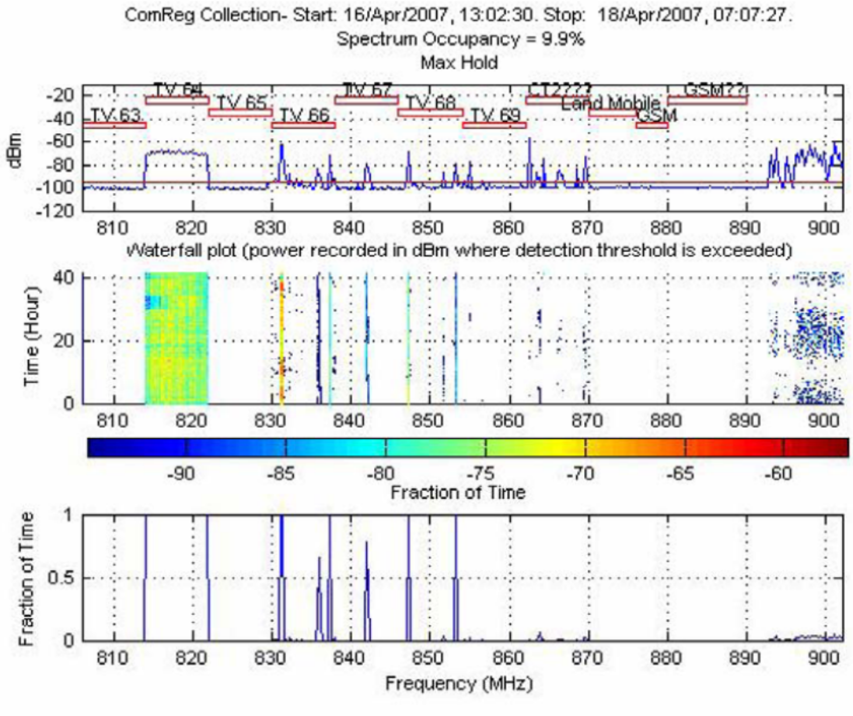
On top of that, the spectrum in pipeline is also much less when compared to that of China. It is to be noted that spectrum in pipeline refers to that band which is not currently available for commercial services but will be made available within three years by the respective governments. Considering such acute spectrum shortage, even the service providers are also finding it increasingly difficult to provide reliable and QoS aware services to these users. This is clearly witnessed in the amount of spectrum subscribed per user which is much less in India (0.2 Hz).

Thereafter, spectrum allocation is mapped with respect to the commonly deployed bands. Based on the obtained data from FCC, it is observed in Table 2.2 that severe spectrum scarcity is experienced in India under the commonly deployed bands when compared to international standards.

Table 2.2 Spectrum availability in commonly deployed bands [2.18]

Band	Europe	USA	India
900 MHz	70	64	12.4
1800 MHz	150	130	97.6
2/2.3 GHz	120	90	60
2.6 GHz	190	194	20
800 MHz	60	70	27.5
Total	590	548	217.5

Interestingly, the flip side of the coin reveals an all-together different perspective of this spectrum shortage problem. While wireless cellular networks face this increasing menacing problem of spectrum congestion, significant portions of the spectrum in other frequency bands (for example, television broadcasting bands) remain vacant for different amounts of time [2.19, 2.20]. This is attributed to the fact that even licensed frequency bands remain unused and unutilized for long durations. Thus, the spectrum scarcity problem can mostly be credited to the inaccuracy in providing efficient spectrum allocation policies, which actually lead to its underutilization [2.13]. To support this claim, spectrum usage measurements were taken in Dublin across different frequency bands as a case-study. It has been observed (Fig. 2.3) that at the most, only 20-25% of spectrum remains efficiently utilized [2.21].



806 MHz to 902 MHz

(a)

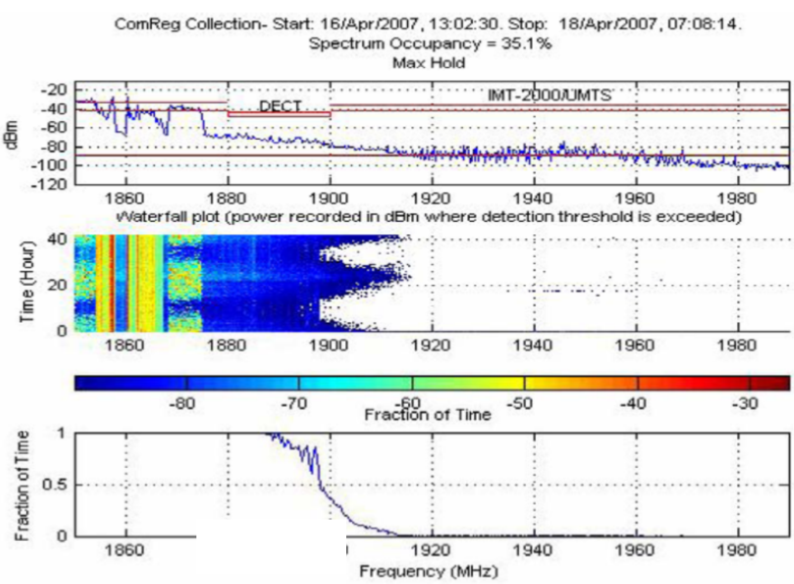
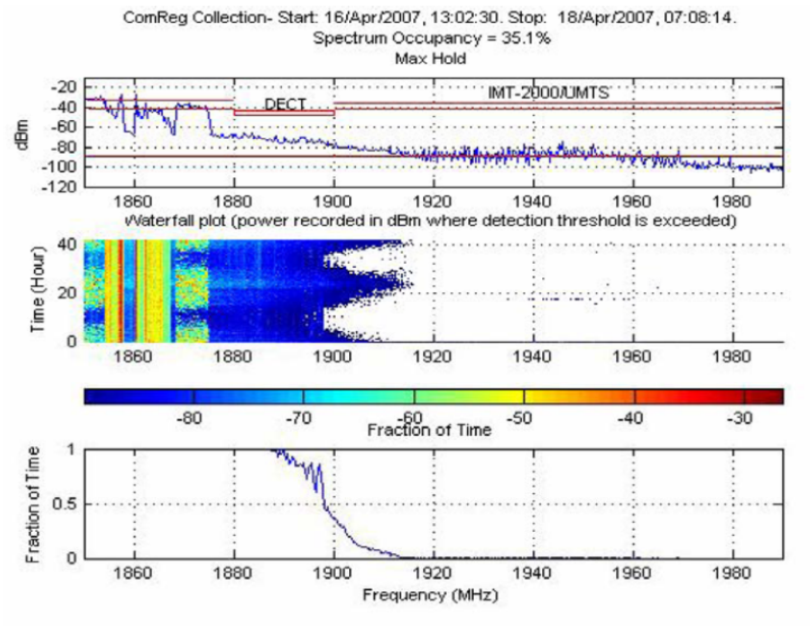


Figure 27. 1850 MHz to 1990 MHz

(b)



1850 MHz to 1990 MHz

(c)

Fig. 2.3 Spectrum Occupancy readings from experimental setup in Dublin, 2007 with respect to different frequency bands in (a), (b), and (c) [2.21].

This implies that the traditional "command and control" approach of spectrum allocation needs a major revision. Under this policy, spectrum allocation is based over frequency and location. That is, the wireless users that are close together in a physical location must be separated by frequency, whereas, the widely separated users reuse the same frequencies. Under the new recommendations, it is proposed that if two users can be properly coordinated, they can share the same frequency in the adjacent locations without experiencing any noticeable interference [2.13]. By enabling these devices to sense their spectrum environment and coordinate their transmissions, more radios can share the same spectrum leading to a rise in spectrum utilization.

This has triggered intensive research towards building next generation wireless networks where users can sense the medium and subsequently deploy cognitive mechanisms to efficiently utilize the spectrum without any harmful interference. Software Defined Radios (SDRs) [2.5, 2.22] are envisioned to play a crucial role towards this development. In such platforms, the radio functionality is primarily embedded in the software rather than the hardware with the advantage that the single radio based user terminal can implement different operational modes and host a multitude of services. By further incorporating cognitive features in such terminals, it is possible to deploy intelligent spectrum management policies leading to a drastic increase in overall spectrum utilization. This has finally led to the emergence of CRN [2.23] where the CR users intelligently tune their reconfigurable and adaptable cognitive radios towards performing medium sensing and transmission operations.

2.1.3 Contributions of this Chapter

As both VoIP and CRN have established their strong presence in the respective domains, this chapter provides a background study of these two technologies. The significant contributions are discussed as follows.

1. Section 2.1 has already introduced VoIP and CRN technologies that form the basis of this thesis work. In this regard, the evolution of VoIP and CRN technologies is traced from the initial days of telephony to the modern era of infrastructure convergence that is also exposed to the potential crisis of spectrum scarcity. In this context, this section points out the enormous significance that these two technologies promise to hold in the current and upcoming wireless communication networks.
2. Section 2.2 provides the technical overview of the VoIP technology and also discusses its relevance in modern day wireless communications. In doing so, the benefits of VoIP are discussed along with the key application areas where VoIP has registered tremendous success.
3. This is followed by the discussion on the role played by CRN technology in the next generation wireless communications in Section 2.3. After providing a brief overview of the "cognitive" aspects of such users, the CR cycle and the CRN architecture are described in detail. Next, the applications that stand to benefit from the operational modes in CRN are highlighted. Finally, the potential design challenges are suitably addressed.
4. After describing the technical aspects pertaining to VoIP and CRN, onus is now on the integration of these two technologies in the context of next generation communications and this issue is aptly taken up for discussion in Section 2.4. The basic system model for supporting VoIP sessions over

a generic CRN is presented. Accordingly, the relevant application areas are highlighted where VoIP services over CRN can be suitably deployed.

5. The mathematical framework enabling the transmission of VoIP packets over CRN is discussed in Section 2.5. VoIP traffic is modeled as an on-off model and matched with the MMPP model using IDC Matching technique. The basic expressions for important system parameters are derived subsequently. In addition, the wireless channel in CRN is modeled as a two-state Markov Chain and its transition from being occupied by PUs to staying unoccupied (and ready for SU arrival) is duly mapped in this section.

6. Section 2.6 presents the key QoS metrics and CRN based system parameters. The primary metrics defining the call quality in VoIP such as delay, jitter, packet loss, MOS and R-Factor are defined and their threshold values are also given for future reference. Likewise, the key system parameters with respect to CRN such as system capacity, spectral efficiency, energy efficiency, spectrum handoff delay and CR timing cycle are also described briefly. At last, the Quality of Experience (QoE) metric is discussed that measures the overall end-user satisfaction during VoIP communications.

Finally, the chapter is concluded in Section 2.7.

2.2 Relevance of VoIP Applications and Services in Wireless Communication

VoIP is a technology that enables the routing of voice communications through Internet or any other IP based networks [2.8]. Here the voice is transmitted over a general purpose packet-switched network instead of dedicated traditional circuit-switched voice transmission lines. The primary advantages that are realized through this implementation include (a) significant savings in network maintenance and operational costs and (b) rapid rollout of new services. Accordingly, this section provides a brief overview of this VoIP technology with its benefits and applications and also establishes its relevance in modern-day user-friendly communications.

2.2.1 Overview of VoIP technology

VoIP communication is characterized by telephone networks that can be interconnected with Internet. The basic technology consists of digitizing the analog voice and sending it in the form of IP packets over the Internet or any other IP based network. An audio input device, such as a microphone, is required at the sending end. The audio signal as recorded by the input device is sampled at a very high rate (at least 8,000 times per second or more) and transformed into digital form by an Analog-to-Digital (A/D) converter. The digitized data is further compressed into very small samples that are collected together into larger chunks and placed into data packets for transmission over the IP network. This process is referred to as packetization. Generally, a single IP packet will contain 10 or more milliseconds of audio, with 20 or 30 milliseconds being the most common instances. There are a number of ways to compress this audio, the algorithm for which is referred to as a "compressor/de-compressor", or simply Codec [2.24]. Many codecs exist for a variety of applications (e.g., movies and sound recordings). With respect to VoIP, these codecs are optimized for compressing voice, which significantly reduce the bandwidth used (compared to an uncompressed audio stream) and ensure the high quality of VoIP transmissions. Most of the codecs are defined by the standards of the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) [2.25]. Each of them has different properties regarding the amount of required bandwidth and the perceived quality of the encoded speech signal.

After the binary information is encoded and packetized at the sender end, packets encapsulating the voice data can be transmitted on the network. Voice packets interact in the network with other application packets and are routed through shared connections to their destination. At the receiver end, they are decapsulated and decoded. The flow of digital data is then converted to analog form again and played at an output device, usually a speaker. Fig. 2.4 shows the end-to-end path as needed for VoIP communication (a similar path exists in the opposite sense for a bi-directional connection).

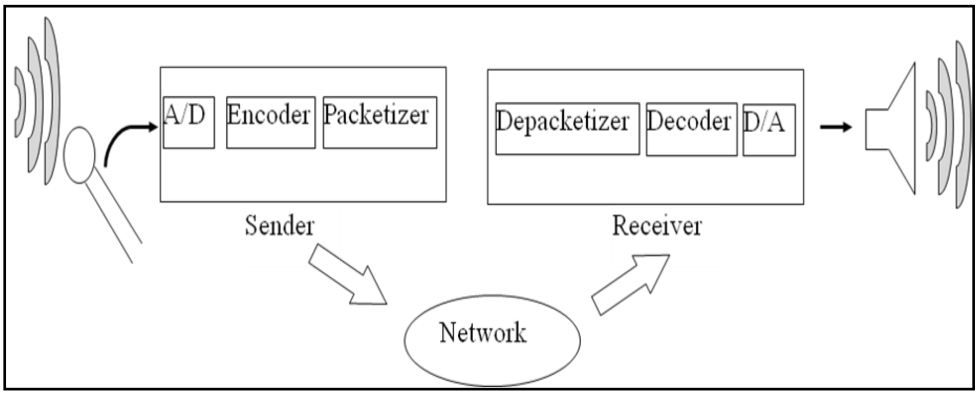


Fig. 2.4 VoIP Technology

However, some IP packets can be lost in transit, especially when the Internet is a typical "best-effort" network. As real-time communication is highly sensitive to such loss of information, steps must be taken to minimize the packet loss through reservation of resources and other techniques. Codecs can compensate for these lost packets by "filling in the gaps" with audio that is acceptable to human ear. This process is referred to as Packet Loss Concealment (PLC) technique [2.26]. Redundancy is another strategy where the packets are sent multiple times in order to overcome packet loss. Error recovery techniques like Forward Error Correction (FEC) include some information from previously transmitted packets in subsequent packets. Thereafter, the lost packet is reconstructed from the information bits in neighboring packets by applying suitable mathematical operations in a particular FEC scheme.

However at times, the packets considered lost may actually reach the destination after incurring a significant amount of delay. It is very common for applications to experience the out-of-order packets in a packet switched network. This is particularly problematic for VoIP systems, as delays in delivering a voice packet means the information is too old to play. Such old packets are discarded in small amounts by the PLC algorithms and perceived as being lost in the network.

2.2.2 Network Architecture

The fundamental elements required to deploy VoIP in a public network are enlisted as follows.

1. IP enabled workstation – The end-users that are involved in VoIP communication must have IP enabled devices that are compatible with VoIP protocols and allow routing through IP based networks. The workstation can be a soft-phone installed in a computer with access to any IP network. IP enabled mobile and fixed telephones can also be used to implement VoIP technology.
2. VoIP Server – The VoIP Server is the centralized node that initiates, manages and terminates communication between the caller and the callee. (In telephony terms, caller is the user who initiates the call and a callee receives the call at the other end). The VoIP Server must implement the call signaling protocols (SIP – Session Initiation Protocol, H.323, etc.) and ensure proper routing of IP packets to their destination. Call admission control is one of the primary functions of the server. It can also be used for QoS provisioning mechanisms,
3. Gateway – One way to increase the interoperability of VoIP is by implementing it in diverse networks having different characteristics. This is made possible by using gateways. Gateways ensure proper coordination in between these networks and further allow VoIP users to communicate to PSTN based telephones. In addition, firewalls can be implemented with appropriate packet filtering rules in the gateways to achieve secured communications.
4. Gatekeeper – A gatekeeper is a management tool that oversees authentication, authorization, telephone directory and PBX services. Commercial entities implementing VoIP can also maintain the billing information along with the call details in the gatekeeper. Although it can be incorporated in the server, generally gatekeepers are implemented separately to simplify the server operations.

An architectural setup for VoIP based communication system comprising of the above entities is demonstrated in Fig. 2.5 that depicts the general deployment of VoIP services in various IP and PSTN based networks.

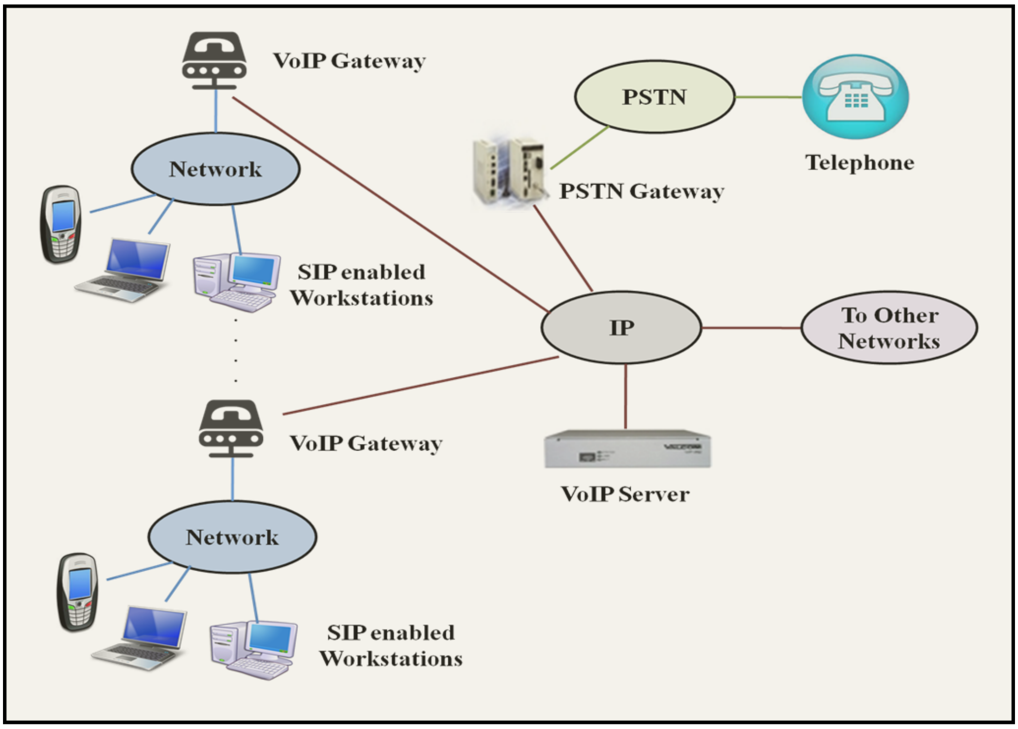


Fig. 2.5 Network Architecture of VoIP based Communication System

2.2.3 VoIP: Why implement it?

VoIP was initially presented as a technology that could enable a service provider to transport voice for "free" over the Internet as transport of packets over IP network was free. Gradually, it found applications in residential telephony as well as in office networks. The success of VoIP can be attributed to the following key reasons.

- ***Ease of deployment*** - Many functions requiring multiple distributed points of presence can be centralized in VoIP domain owing to the unique characteristics of VoIP call-controllers, thereby reducing administrative overheads and accelerating deployment.
- ***Simplification of transport networks*** - Standard IP networks after proper configuration can be used to carry VoIP packets, thus eliminating the need to establish leased lines dedicated to voice prior to establishment.
- ***Cost reduction*** - There is significant reduction in operational and maintenance costs. This is especially beneficial for companies that

actively make a lot of calls on a daily basis, or for those who execute long-distance international calls.

- ***Value added services*** - VoIP infrastructure can be utilized to host and implement various services for consumers like Multimedia Messaging Service (MMS), Push-To-Talk (PTT), etc.
- ***Anytime anywhere communication*** – IM-based VoIP (as depicted in Table 1.1) offers anytime anywhere communication to customers having access to Internet and a registered account, thereby removing the problems of infrastructure-based modes of communication.
- ***Easy Upgradation*** - VoIP services can be easily upgraded owing to the simplicity of VoIP operations.

However, as VoIP operates over IP which is the "best effort" protocol, it requires certain QoS guarantees [2.26]. The real-time loss sensitive nature of voice communication implies that extreme care must be taken towards maintaining the quality of call that is acceptable to the end-user. Security concerns should also be addressed since voice packets can easily be compromised and private sessions can be hacked.

2.2.4 VoIP Applications

Apart from its dominating presence in IP telephony, VoIP can be used to develop several other applications. As telecom operators are focusing on value added services in order to attract more customers, VoIP technology serves as an excellent platform to design such tools. This is only made possible by "infrastructure convergence" of VoIP with data networks, which provides a low-cost easy solution to many real world problems. With the emergence of simple text based protocols like Session Initiation Protocol (SIP) for VoIP, existing applications can be easily modified to enjoy the benefits of all-IP communication.

One of the most popular applications in mobile telephony is SMS (Short Message Service) [2.27] that uses standardized communication protocols to exchange short text messages between fixed line or mobile phone devices. MMS [2.27] extends the messaging service to allow simultaneous exchange of text, audio and video files. It has been observed that there are several disadvantages with messaging services in traditional GSM (Global System for Mobile Communications) networks. The primary drawback is on the limit of messages that a single GSM modem can handle. The situation is further worsened with MMS messages that contain media files and consume higher bandwidth. Even though the capacity can be increased by using more modems, the connection fails during increased traffic activity as witnessed during the New Year Days and other public holidays. Moreover, only a fixed sender address can be used to send messages.

All these problems can be effectively reduced by implementing IP based messaging using the existing VoIP infrastructure. Firstly, any IP enabled device can send the messages. Thus, users can transmit messages during emergency even in the absence of a phone or GSM connection. Moreover, low signal strength would not affect IP message delivery. Finally, the same IP device can be used by different senders to forward messages using their own registered accounts. As per industry reports, the mobile industry reported losses of over US$ 10 billion due to declining text messages sent by users as they switched to IP based messaging service.

Multi-party conferencing applications are also relying on VoIP so that professionals from across the business and service sector can instantly connect among themselves “on the fly”. In addition to the cost-saving benefits derived from using VoIP based conferencing services, users also can take advantage of its additional facilities such as checking the voice mail over the Internet, attach messages to email and sharing files during conversation. An overview of the VoIP based multi-conferencing system is depicted in Fig. 2.6. The VoIP conference server performs authentication and authorization to allow only registered users in the conference. Call details along with the necessary statistics are also gathered for accounting purposes.

Moreover, there has been an unprecedented growth in the popularity of social networking applications with the emergence of Facebook, WhatsApp, Linkedln, Google Talk and other popular web-based services that allow people from around the globe to connect and share ideas. These applications provide users with their registered accounts to meet other people having similar interests and interact via text, audio and video chat. As most of these service providers operate without charging any money from the users, VoIP is the best option to implement the required facilities.

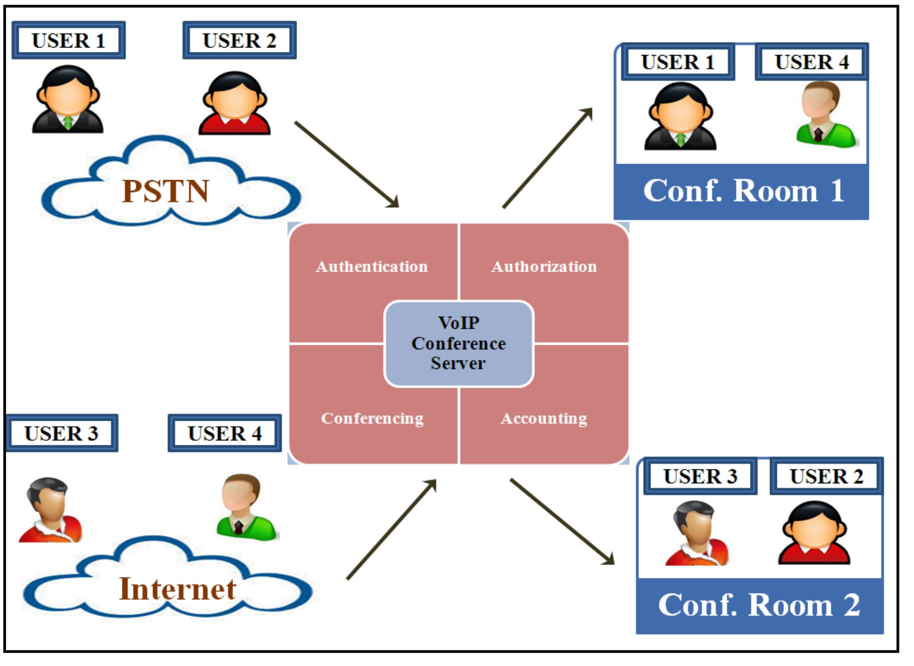


Fig. 2.6 Role of VoIP in multi-conferencing system

Multiplayer gaming is another domain where VoIP has a bright prospect [2.28]. Games, nowadays, give players realistic models, believable backgrounds and the ability to use the Internet to connect to millions of other gamers all over the world. Allowing multiplayer gamers to cooperate and play has been made possible with VoIP, which, despite its shortcomings, has made playing such games more exciting. The primary advantage lies in the seamless integration of the gaming software with VoIP applications, thereby allowing the concept of rapid immersion where users do not have to switch to separate windows to share information with fellow gamers. VoIP also enables users to invite players into the game at runtime without any interruption, thereby retaining the excitement level in these games and expanding the popularity of the gaming applications.

Currently, military organizations are also transitioning their telephony infrastructure from legacy Time Division Multiplexing (TDM) based operational modes to Next Generation Networks (NGN) based on VoIP technology [2.29]. Other than the basic advantages of all-IP based VoIP communication, VoIP proves to be more resilient than TDM networks and easier to manage compared to their older TDM counterparts. Security and survivability are obvious military requirements that must be satisfied by VoIP applications. Accordingly, military VoIP networks are based on modified versions of the standard protocols. One example is MLPP policy (Multilayer Precedence and Preemption) that is an ITU defined standard [2.30] being used by US Department of Defense and provides a prioritized call handling service having different precedence and preemption features.

Thus, VoIP continues to mark its growing presence in almost every sector that involves IP enabled services and is going to enjoy a significant share of the total web based traffic in the near future.

2.2.5 VoIP Popularity

It is already evident from the discussions that VoIP services have attracted vast attention from industries and individuals alike. On a quantitative note, the advantages of VoIP can only be realized through increased usage of Internet. As per ITU-T data, recent statistics record higher usage of Internet by consumers across the globe that has reached almost 90 percent. This implies the huge prospect of Internet based VoIP services in these countries. Even developing countries like India and Malaysia have witnessed greater utilization of Internet owing to higher literacy rates and increased availability of Internet based connectivity as depicted in Fig. 2.7.

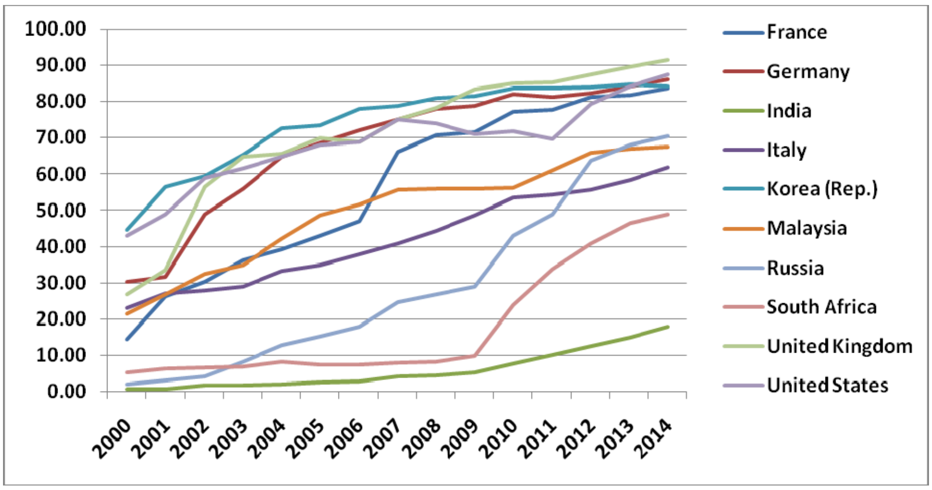


Fig. 2.7 Percentage utilization of Internet by users in various countries [2.31]

Additionally, the total number of mobile subscribers has increased drastically in all countries, more specifically in the developing sector. At the same time, the number of people opting for mobile broadband subscriptions has increased manifold over the past few years as depicted in Fig. 2.8. This trend highlights the growing dominance of mobile based IP services and underlines the significance of the VoIP as the primary communication medium in this domain.

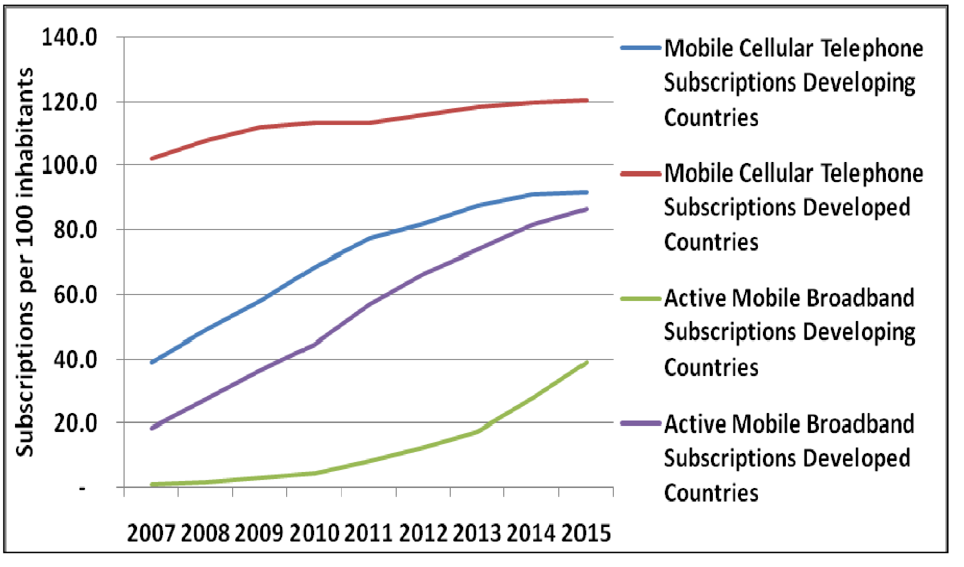


Fig. 2.8 Mobile Telephony and Mobile Broadband users in developing and developed countries [2.31]

Quite obviously, the popularity of VoIP both as a service and as an application has gained immense momentum in the recent years, resulting in high volumes of VoIP traffic generated as shown in Fig. 2.9. The industry success is also stimulated by the US FCC decision (not to control or limit voice traffic over the Internet) and the cheap price of these services. With low barriers to entry, competition is growing making the companies differentiate the services, improve the call quality and reduce the subscription prices.

As per reports published in Point Topic [2.32], VoIP has witnessed a phenomenal growth in terms of subscribers (as illustrated in Fig. 2.10) that can be attributed to the emergence of professional VoIP services along with increased deployment in campus and office networks. The primary reason behind this surge is not only the cost-effective solution of VoIP. Rather, it also includes choosing the right telephony solution that will provide "exceptional customer experience" and boost its perception, which is where VoIP excels over traditional telephony solutions. Moreover, increasing number of households especially in the developed countries of US and Europe are switching to VoIP service from traditional landline telephony due to its easy integration with other interactive services.

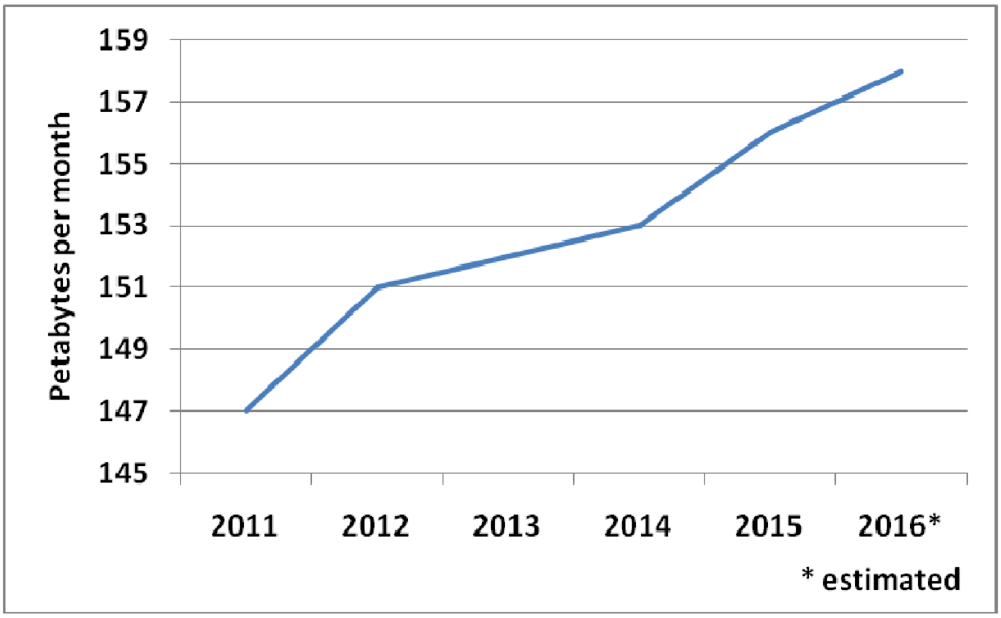


Fig. 2.9 Volumes of VoIP traffic generated over the years [2.31]

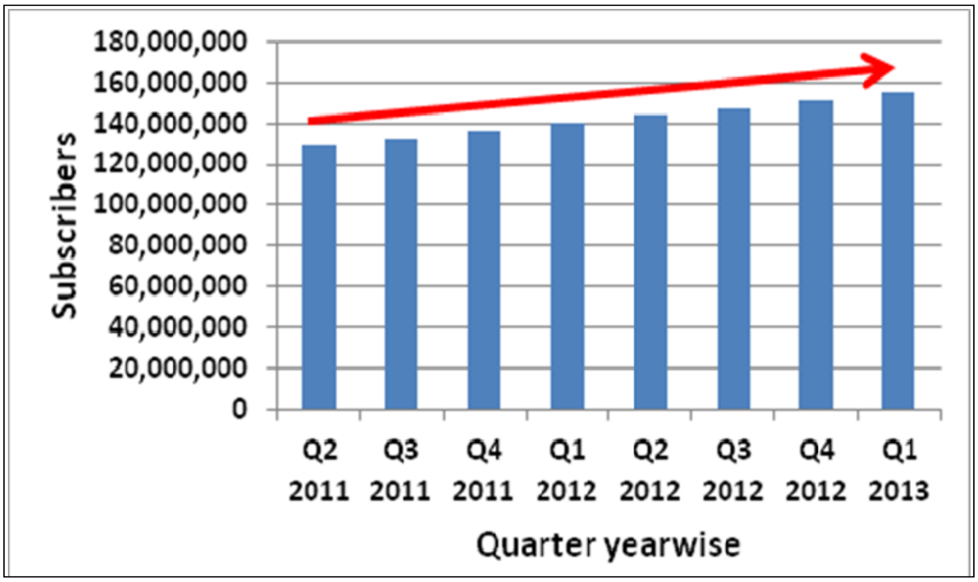


Fig. 2.10 Total number of VoIP subscribers for various quarters of 2011-2013 [2.32]

The continent-wise distribution of the VoIP subscribers is depicted in Fig. 2.11. It is evident that VoIP services are most efficiently utilized in the highly developed areas of North America and Europe. Countries like Korea Republic and Japan in the East Asia also enjoy a significant share of the overall VoIP usage. Specifically, the United States of America became the country with the highest number of subscribers in 2013. During the same period, France recorded the maximum percentage growth where almost 95% of the overall number of fixed broadband users subscribed for VoIP in the country.

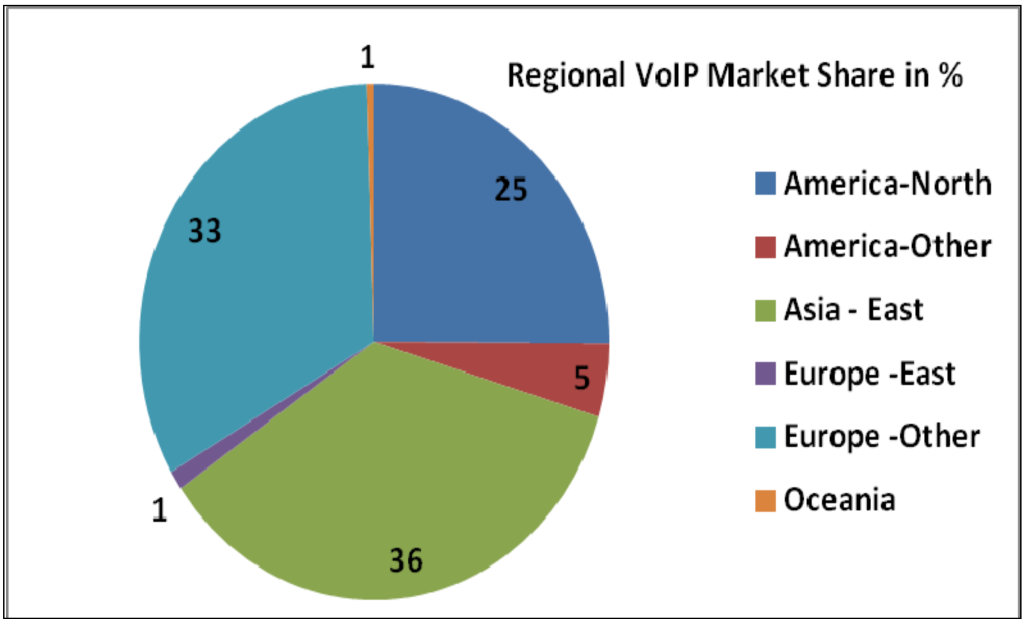


Fig. 2.11 Distribution of VoIP users across continents [2.32]

Focus is now shifted from the VoIP applications to the CRN technology which serves as the basic network platform in this thesis and is discussed in the next section.

2.3 Emergence of Cognitive Radio Networks

Traditionally, the Fixed Spectrum Access Policy (FSA) [2.33] has been adopted by spectrum regulators to host different non-interfering wireless services. In this policy, certain portion of the spectrum is allotted to one or more registered users who possess the license to transmit in those bands. Thus, this spectrum is dedicated to only the licensed users irrespective of whether these users are using this band or not. This has, over the years, resulted in a severe crisis of the available spectrum (as already highlighted in Section 2.1). At the same time, rigorous studies on spectrum survey [2.20, 2.21, 2.34] have pointed out to large portions of vacant spectrum being under-utilized. This obviously points to the failure of the spectrum allocation mechanisms to optimally use spectrum as a resource, resulting in its uneven distribution and contributing to the spectrum congestion problem even though the spectrum remains physically available.

Therefore, the need of the hour is to devise efficient spectrum allocation policies that will reduce the spectrum scarcity crisis, while also increase the overall spectrum utilization. This, in turn, has led to the development of the Dynamic Spectrum Allocation (DSA) policy [2.5, 2.33] where the spectrum is no longer used solely by the dedicated users. Rather, the licensed users obtain the topmost priority to use the bands. However, in their absence, this spectrum is used by the unlicensed (or unregistered) users. While the licensed users are denoted by the Primary Users (or PUs), the unlicensed users are termed as Secondary Users (or SUs). The SUs can use the licensed spectrum in collaboration with the PUs either by transmitting in their absence or using shared mechanisms. In either case, the protection of the PUs demands the highest priority. Therefore, these SUs must sense the wireless channel for PU activity and accordingly take cognitive decisions regarding optimal usage of the licensed spectrum. Thus, the SUs are modeled as Cognitive Radio Users [2.22] and the overall network is termed as the Cognitive Radio Network (CRN).

Two types of spectrum access are proposed for the SUs in CRN. The first policy is the Opportunistic Spectrum Access (OSA) [2.35] where the SUs can access the spectrum on a temporal basis provided that the PUs are absent. The second policy is the Concurrent Spectrum Access (CSA) [2.35] where the SU transmits with refrained transmission power so as to maintain the interference with the primary receiver below a certain threshold level. Both these policies are depicted in Fig 2.12. OSA and CSA are also denoted by CR overlay and underlay schemes respectively.

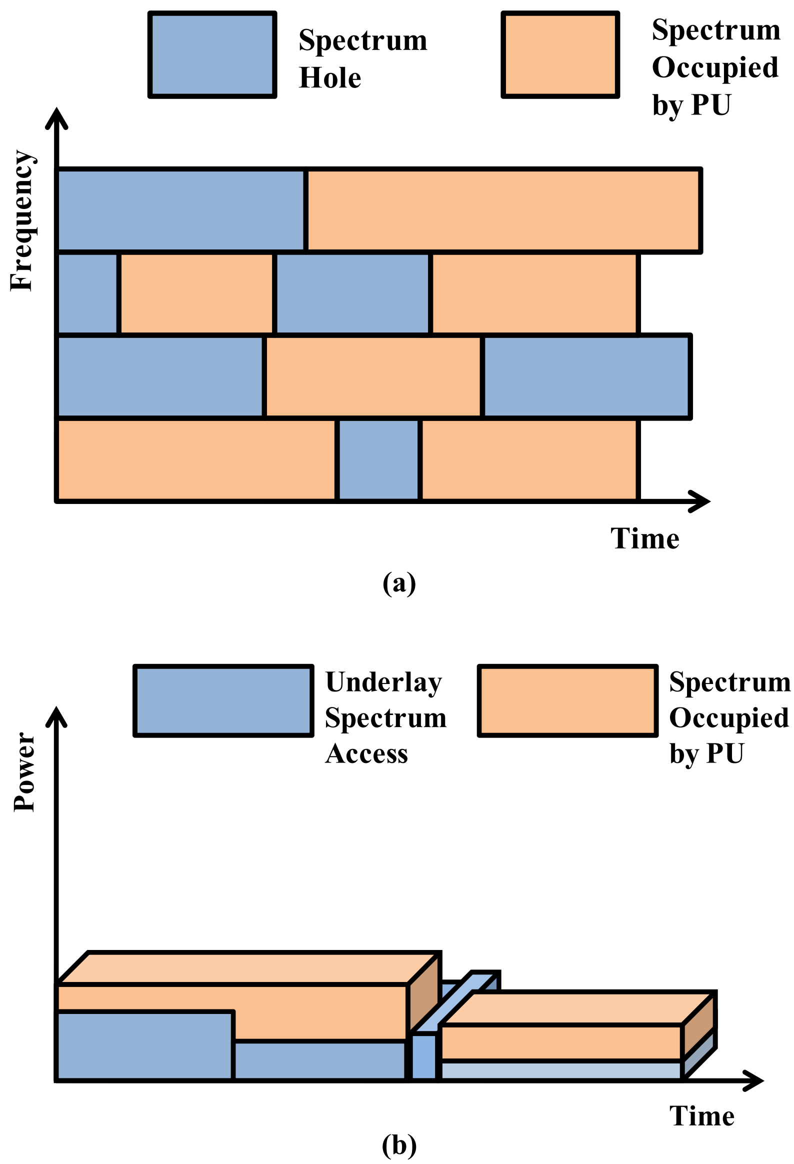


Fig. 2.12 Dynamic Spectrum Access in CRN with respect to a) overlay policy, and b) underlay policy

2.3.1 Who is a CR User?

The first question that arises with respect to CRN is the classification of a CR user and how it is different from a normal user. Precisely, any SU admitted in the CRN becomes a CR user when it exhibits two unique capabilities [2.22].

- ***Cognitive Capability***: Here the SUs are equipped with a special transceiver (also termed as Cognitive Radio or CR) that can sense the spectrum band and using sophisticated techniques determine the spatial variations in the radio environment. Based on the outcome, the user can determine which portions of the spectrum band remain idle at a specific time or location.

- ***Reconfigurability:*** This property when possessed by the CR user makes its radio dynamically programmable depending on the current network conditions. In particular, the CR user must be able to vary its operating characteristics at run-time (including operating frequency, transmission power, communication technology, modulation schemes, etc.) based on the outcome of its decision mechanisms. Obviously, this must be supported by the underlying hardware. One way to achieve this reconfigurability feature is through the use of Software Defined Radio (SDR). Here the radio terminals can be configured on the run using software codes that can configure its characteristics as and when the situation demands.

Equipped with these two unique features, the CR user displays its cognitive capabilities and follows the basic CR Cycle which is discussed in the next section.

2.3.2 CR cycle

The basic CR cycle is depicted in Fig 2.13. Every SU follows this CR cycle with minor modifications and performs the following essential operations.

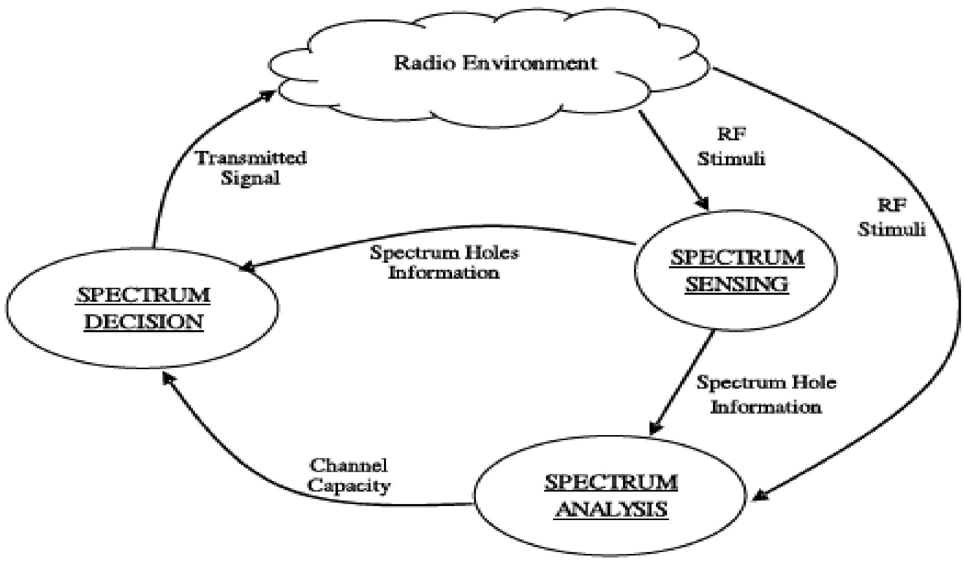


Fig. 2.13 Basic CR Cycle [2.5]

1. Spectrum sensing: This is one of the most critical aspects of CRN, where the CR detects the spectrum holes through periodic sensing mechanisms (energy detection, matched filter detection, cyclostationary feature detection etc.) and determines the suitability of a particular wireless channel for hosting SU applications. It is imperative that higher the accuracy of sensing, more is the success rate in terms of increasing system capacity and subsequent spectrum utilization. On the other hand, imperfect spectrum sensing can lead to the undesirable conditions of false-alarm (where channel is detected busy even though it is idle) and miss-detection (where channel is detected idle when it is actually occupied by PU). Both of them can severely degrade the SU performance in terms of system under-utilization as well as increased levels of harmful interference with the PUs.
2. Spectrum analysis: This step follows the spectrum sensing part, where the SU uses the sensing outcome to analyze the channel conditions before considering the wireless channel as an “opportunity” for transmission. This is important because considering the unreliability of the wireless environment, the sensed channel must be analyzed and characterized in terms of different factors (including interference levels, path loss, wireless link errors, link layer delay, holding time of the PUs, etc.). Based on this analysis, the SU enters the third phase.
3. Spectrum decision: Here the SU takes any of the three decisions namely; i) the decision to transmit in the current channel considering favorable channel conditions for the underlying SU transmissions; ii) the decision to halt transmission when the current channel is analyzed to be unsuitable for sending SU packets. This is followed by the decision to perform spectrum handoff to another available idle channel; and finally, iii) the decision to drop from the channel when the operating channel is inapt for SU transmissions and no other idle channel is available at that time instant.

Thus, spectrum decision determines the actual transmission by the SU. Also, it includes another important aspect of spectrum management, namely spectrum mobility which involves selecting a suitable target channel and performing spectrum handoff to that channel when PU arrives in the current operating channel.

Collectively, spectrum sensing, spectrum analysis and spectrum decision including spectrum mobility constitute the CR cycle. Thereafter, when these SUs are deployed in the network along with the PUs, the network becomes a Cognitive Radio Network. The resultant network architecture is discussed as follows.

2.3.3 Cognitive Radio Network Architecture

CRN comprises of several PUs and SUs that operate using the DSA policy as described in the previous section. This CRN may be infrastructure-based network or an ad-hoc network.

In the infrastructure-based network, the CRN comprises of the SUs, the SU base-stations and the Spectrum Broker (also denoted by Spectrum Controller or SC) node [2.5]. These base stations provide single hop connection to the SUs by means of which these SUs can access other networks. The spectrum broker is unique to the CRN deployment. It is the central resource allocation and management entity that shares the available licensed spectrum among different networks involving the SUs and plays a crucial role towards ensuring proper usage of the under-utilized spectrum.

In the ad-hoc network, the SUs operate independently, either non-cooperatively [2.36] or in collaboration with each other [2.37, 2.38]. There is no centralized entity to control their actions. As with any ad-hoc network, the efficiency of the ad-hoc CRN depends on the timely decision- making capabilities of these SUs, which, unless properly planned, can severely restrict the overall system capacity of the network.

Overall, the general CRN architecture is depicted in Fig 2.14 [2.5].

2.3.4 Applications of CRN

As CRN is envisioned to play a critical role in the next generation wireless communication [2.39], this section highlights the different applications where the benefits of CRN can be effectively realized.

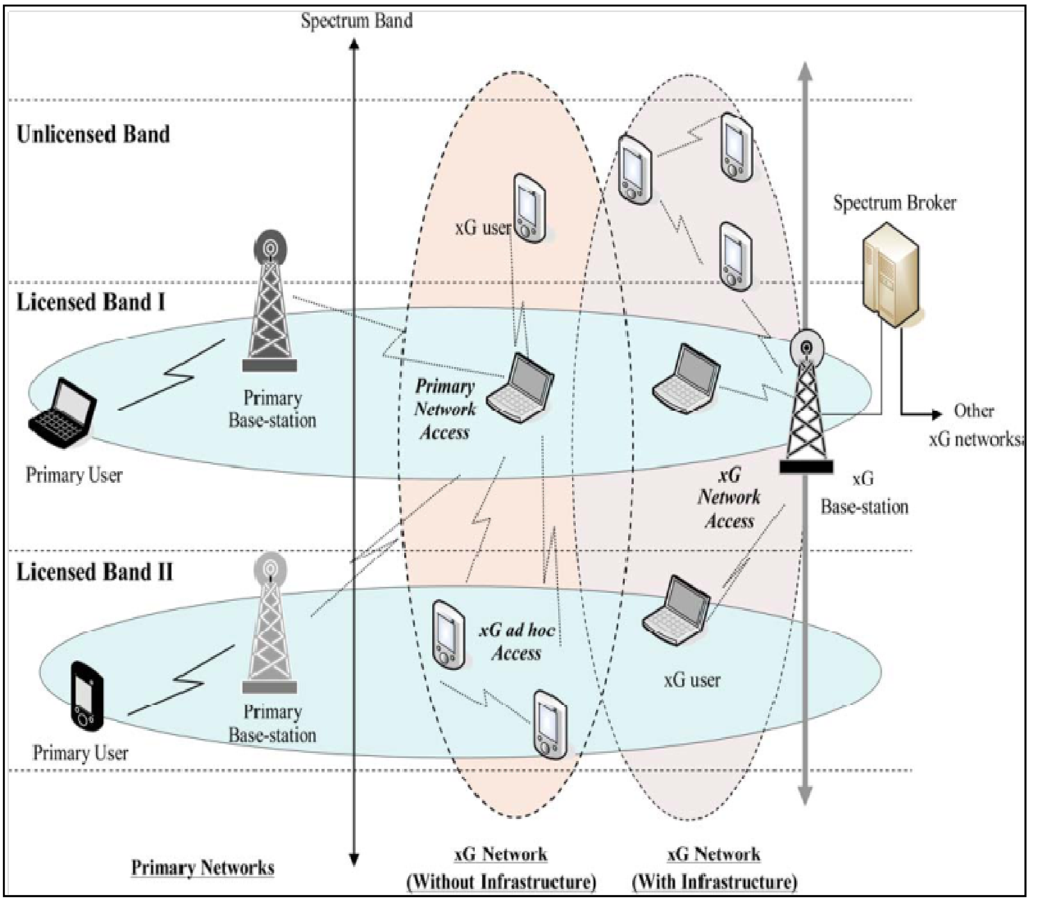


Fig. 2.14 CRN Architecture [2.5]

- ***Vehicular Networks:*** One of the interesting applications of CRN can be found in the area of vehicular communication [2.40]. This is particularly relevant in modern society, given the state-of-art research with respect to in-car entertainment services and public safety systems that have gradually evolved over time. The CR enabled vehicles are required to opportunistically utilize the spectrum (specifically the digital television frequency bands in the Ultra High Frequency (UHF) range) and establish the links for data transmission on the go. However, this does not merely refer to placing a CR enabled device inside the car. Rather, the CR based vehicles actually detect spectrum holes, analyze them, select suitable frequencies and subsequently transmit over them without causing harmful interference to the licensed users in that zone. The fundamental difference with the conventional CR systems is that unlike those systems, the spectrum availability in CR enabled vehicular communication varies dynamically with time and is a function of not only the PU traffic activities, but also the relative motion between the PUs and the vehicular SUs. Such CR based vehicular networks can be designed under three categories – i) Vehicles rely on cooperation to increase their sensing accuracy and effectively utilize the spectrum holes; ii) Vehicles periodically communicate with the road-side Base Stations (BS) and uses the repository of data stored in these BS for utilizing the vacant spectrum.; iii) A completely centralized system is envisioned where the BSs autonomously allocate channels to the vehicular SUs without relying on information from them.

- ***Underwater Acoustic Communications:*** Underwater Acoustic Communication (UAC) is another emerging domain that is poised to deploy CR services pretty soon. Typically, the UAC based systems are severely constrained by the scarcely available acoustic spectrum that is prone to high levels of path-loss and noise. Therefore, sharing this spectrum among different underwater systems (underwater vehicles, fleet, acoustic sensor networks for underwater exploration, etc.) is a challenging task, given the fact that this spectrum exhibits spatio-temporal variations in its characteristics. For example, the path-loss of the medium changes with depth and seasons, the noise is amplified by waves and human activities, and the nearby active sonars can adversely affect the environment. Thus, CRN in the form of DSA (and OSA) can prove to be immensely useful for underwater communications. Such communications can be termed as Cognitive Acoustic Communications [2.41]. In such networks, the CR empowered underwater nodes can sense vacant acoustic spectrum and accordingly configure their transmission properties to minimize any interference with the other existing underwater devices or sonars.
- ***Smart Grid:*** Another innovative application of CRN is in the area of smart grids. It has already been observed that increasing population coupled with higher demands for power have put severe constraints on the existing power grids, as a result of which system reliability, power quality and customer satisfaction have witnessed a gradual decline over

the years. In addition, the harsh environmental conditions surrounding these power grids including dynamic topology changes, interference and fading concerns, and connectivity issues have issued challenging problems in relation to establishing wireless communication in these grids. Consequently, this has led to the development of CR enabled next-generation power grids, commonly referred to as "Smart Grids" (SG). These SGs are the modernized versions of the traditional power grids and comprise of the advanced ICT infrastructure. CR technology further aims to increase the system capacity by addressing the challenges of wireless link failures and extreme environmental conditions in these power grids, through the incorporation of DSA based spectral management policies [2.42]. For example, CRN can make provisions for large-scale SG systems to use different spectrum regions and also enable fair sharing of spectrum resources among the various subsystems within a large SG system. In this regard, CRN can be put to good use for several SG services such as, i) advanced metering architecture, ii) distributed electric generation, iii) power outage detection, and iv) wide area monitoring.

- ***Emergency and Public Safety Communications:*** One of the primary driving factors towards developing CRN on a large scale basis is the availability of the unused TV frequency bands. CR deployment over such bands has been officially termed as TVWS (TV White Spaces) [2.43]. Apart from rural and broadband home wireless services and vehicular and smart grid communications, TVWS also finds its application in another important domain – the Portable Cognitive Emergency Networks. The advantage of VHF-UHF bands in TVWS over Gigahertz bands in terms of wider coverage and higher signal penetration can be effectively utilized for rescue operations and communications with the centrally located service centers. The CRN comprises of Mobile Stations (MSs) used by the rescue officials and the Cognitive Base Stations (BS) that are top-mounted on the vehicles. Connectivity between MS and BS is established through the air interface operating in the TVWS zone. As a specific example, in order to locate the survivors under a damaged building in times of disaster such as earthquakes, the higher penetration power of the signals can be put to effective use to trace their presence by requesting for the digital maps and heat signatures from the BS. Likewise after rescue is done, MS can transmit the live data regarding health and vital conditions of the rescued victims to the remotely located medical team using the top-mounted BSs. Recent earthquakes in Japan and other man-made disasters and terrorist activities have accelerated further research in this discipline.

- ***Multimedia Applications:*** The impetus for the design of CRN was greatly influenced by the demand for providing better mobile multimedia services, as proposed by Mitola in his landmark paper titled "Cognitive radio for flexible mobile multimedia communications" where he proposed the usage of spectrum pooling for increased transmission opportunities. Four spectrum pools are suggested in [2.4] that are ideally suited for multimedia applications. Subsequently, the advantages of spectrum pooling are highlighted where different strategies are devised in relation to spectrum sharing among multiple SU subsystems. Thereafter, this concept has been used in several studies such as [2.44] and [2.45] where the SU base stations periodically broadcast detection frames which are used by the SUs for sensing. Using OFDM (Orthogonal Frequency Division Multiplexing), it is possible to feed certain sub-carriers with zeros resulting in zero emission of radio power on the carriers used by the PUs for transmission. As higher data rates and integrated multimedia applications continue to grow with 4G and 5G networks, fulfilling their diversified requirements while still allowing maximum number of users to share the spectrum resource poses an enormous challenge for the service providers, that can effectively be supported by CRN through proper capacity planning and dynamic spectrum management policies.

In addition, the prospects and challenges of CRN have been studied with respect to several other applications such as Wireless Body Area Networks (WBAN) [2.46], Wireless Sensor Networks (WSN) [2.47], etc.

2.3.5 Challenges in CRN

In view of the complexities inherent in CRN [2.5, 2.48], the key challenges in relation to successful design and deployment of CRN are highlighted in this section as follows.

1. ***Sensing Issues:*** The primary challenge lies in the design of the CR module such that it is able to detect the PU presence with minimum complexity and ensure successful transmission by SU (if that channel is idle) without harmful interference. However, in a CRN, a wireless channel may either be occupied by PU or SU. Therefore, spectrum sensing should ideally be able to detect which user is currently occupying it and afterwards determine the channel busy/idle characteristics using estimation and learning techniques. As sensing overhead does lead to transmission halts, the CR must be carefully configured specially when dealing with real-time multimedia traffic such as VoIP. Also, when collaborative sensing techniques are applied, SUs have to ensure that all the user decisions are collectively taken into account before arriving at the final conclusion regarding channel availability and suitability for hosting SU traffic.

2. ***Spectrum Management Issues:*** As the ultimate objective of CRN is to allow transmission by SUs albeit in sharing mode with the PUs, spectrum management can tend to be highly complex unless guided by appropriate reconfigurable policies [2.48]. At first, the spectrum analysis must not be based on SNR (Signal-to-noise ratio) values alone. Rather different parameters like the wireless link errors, holding time by PU, path-loss, fading and shadowing models, etc. must be taken into account during analysis. Next, it must be ensured that when SUs transmit in non-contiguous bands, spectrum decision must decide as to how many bands should be taken account of and how to schedule the transmissions across these bands. Also, the spectrum decision is greatly affected in heterogeneous networks, where different types of SUs exist (for example, RT and NRT SUs). Care must be taken to ensure efficient design of queuing models and call admission control policies to incorporate maximum number of such users in the system without

compromising with the service quality. Finally, spectrum management must also include the necessary framework that will support cooperation among the SUs while allowing the reconfiguration of the underlying CR modules.

3. ***Spectrum Mobility Issues:*** When several channels are available to the SUs, the decision framework must decide the target channels where SUs can perform handoff as and when required. This requires the design of TCS for each SU, which should also be updated with time to capture the variation in network dynamics. Then, the next challenge lies in performing the spectrum handoff without causing serious degradation to the supported applications. This is a difficult task, given the fact that spectrum handoff requires SUs to change their operating frequencies, and involves significant overhead depending on the underlying hardware. It must also be understood that spectrum handoff occurs not only due to PU arrival, but also when SUs move from one cell to another (inter-cell handoff) or when the channel conditions degrade considerably (wireless environmental factors). These conditions must also be incorporated while developing handoff algorithms.

4. ***Spectrum Sharing Issues:*** CRN considers spectrum as a resource and allows its sharing among multiple SUs or SU based systems. As with any resource sharing, cooperation and coordination among the entities during resource usage are the main design challenges. One solution is to use Common Control Channel (CCC) which allows the signaling messages to be sent over a dedicated channel. Use of CCC can enhance the coordination among these SUs both in infrastructure based networks and ad-hoc networks. However, PUs have authority to access any licensed channel and CCC is not exempted from this. Hence, sudden unavailability of this CCC due to PU arrival can lead to synchronization issues among the transmitting users. Thus, CCC must be regularly updated and this updated information should be communicated to the SUs in time. This also leads to a re-design of the conventional transmitter-receiver handshake procedure which now needs to

incorporate additional constraints of CCC unavailability and spectrum handoff operations.

5. ***Layered Model Issues:*** Apart from the MAC layer issues (spectrum sharing, management) and PHY (Physical) layer operations (spectrum sensing, reconfiguration, transmission), challenges exist in the upper layers of the SUs. For example, in a conventional network, when a data path is established in the network layer, it can be used over long durations. However, this is not the case in CRN, where this path may get blocked due to random PU activities. Then, either the transmission needs to be re-routed or else the SUs must wait for the recovery of the original path, which is again a spectrum decision problem. Also, the typical congestion control mechanisms in the transport layer cannot directly be applied to CRN. This is attributed to the unique problem in CRN where it is difficult to determine the exact cause of packet drop, that is, whether packet loss occurred due to congestion or due to emergence of the PUs. One solution is to develop explicit mechanisms where the source nodes can determine the PU activity through detection mechanisms and accordingly control their data traffic rate. Therefore, the layered model puts forth several open research issues with respect to CRN.

6. ***Cross-layer Design Issues:*** It is pretty evident from the background study in this section that cross-layer design is an integral aspect in CRN. The most important one is the cross-layer interaction between the MAC layer (for spectrum management and decision-making) and the PHY layer (for spectrum sensing and transmission operations). The challenge lies in designing the related policies in each layer in such a way that they complement each other. For example, if a dynamic sensing interval is applied in the PHY layer to increase transmission opportunities for SUs, the MAC layer should suitably configure its buffers to store the incoming packets from the upper layers and route them accordingly to the lower layers. Likewise during spectrum handoff, cross-layer interaction is required among the PHY, MAC, Transport and Application layers. To cite an instance, whenever an SU changes its operating frequency based on the handoff design, the same information

has to be used by the Transport and Application layers so as to minimize the abrupt degradation in service quality. Additionally, the route information in the Transport layer may be used for route recovery during spectrum handoff.

Thus it is inferred from this section that unique challenges and constraints are posed by CRN in comparison to traditional wireless networks. On top of that, when application-oriented studies are carried out over CRN (for example, multimedia communications, vehicular networks, military networks, public safety situations), the additional requirements of these applications must also be taken care of. This demands joint formulation of application-oriented spectrum management policies such that both the application objectives as well the fundamental objective of CRN (maximize system utilization and reduce spectrum congestion) are fulfilled. Taking cue from this, the next section presents the application-oriented study of VoIP applications over CRN.

2.4 VoIP Services over CRN: Prospects and Applications

As already discussed in the previous section, the primary objective of CRN is to reduce spectrum congestion and utilize the bandwidth at the maximum. At the same time, considering the immense popularity of VoIP in these years, supporting more number of QoS aware VoIP users in traditional networks becomes impossible. Therefore, increasing demand for VoIP services makes it a suitable candidate to be implemented in CRN [2.9] and is discussed in this section.

2.4.1 Architecture

An overview of the network infrastructure for VoIP in CRN is provided in Fig. 2.15. Here the VoIP calls are initiated and managed by the unlicensed SUs in the CRN.

The key operations are described as follows.

1. SUs request for and obtain default channels (frequency bands) for initiating transmissions in the absence of the licensed PUs.

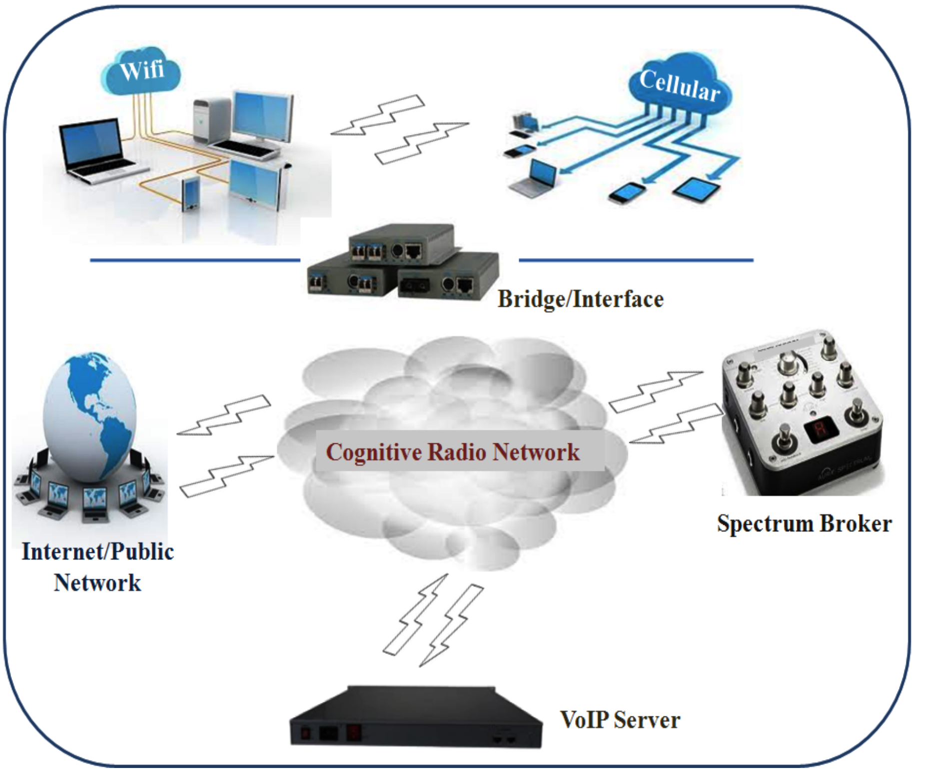


Fig. 2.15 Network Architecture depicting deployment of VoIP services over CRN

2. SUs request for and obtain default channels (frequency bands) for initiating transmissions in the absence of the licensed PUs.
3. Two SUs initiate VoIP communication in the default channel following standard call signaling protocols (SIP or H.323) and start exchanging real-time data in the form of RTP (Real-Time Transport Protocol) /RTCP (Real-time Transport Control Protocol) packets.
4. In between such exchanges, the SUs periodically sense the channel for any PU activity. Three possible cases may occur based on the sensing outcome.
    a. Firstly, if the channel is sensed idle, the SUs can resume their ongoing VoIP session and carry on with the communication.
    b. Secondly, if the channel is sensed busy, the SUs stop their transmission and perform "spectrum handoff" to another available idle channel where they resume the disrupted VoIP call.
    c. Finally, in the third case, if the channel is sensed busy and no other idle channel is available, the VoIP call is dropped by the SUs.

5. After the call is discontinued by the end-user, the termination of VoIP calls follows the standard procedure [2.49] where the underlying call signaling protocol based messages are exchanged between the SU caller and callee and the session is finally closed.

2.4.2 Applications

Once the CRN is successfully poised to host VoIP services, this will usher in a new era of communication mode having enormous significance in the context of several systems, some of which are discussed as follows.

1. ***Disaster management systems:*** Whenever disaster in the form of natural calamities or induced emergencies occurs, communication assumes the foremost priority that will initiate recovery and aid measures. However, in most cases, this communication is severely disrupted in the traditional networks either due to physical (mechanical) damage to the nodal elements, or due to massive traffic load (enormous number of panic calls) on the network elements. VoIP applications over CRN can take advantage of the existing infrastructure and perform a quick setup of VoIP calls for rapid restoration of communication.

2. ***Cellular Communication Models:*** The VoIP based SU terminal can be implemented in the modern cellular phones after suitably configuring their MAC and PHY layers. This will enable these phones to act as SUs and perform VoIP based communication in the idle frequency bands of other service providers, when their underlying cellular network becomes congested. The service providers too can use this platform to provide their unused spectrum to the CR users on a temporal basis and at a nominal price. Thus, this technology will prove beneficial to both the end-users as well as the service providers.

3. ***Service Provider Scenarios:*** As VoIP continues to contribute immensely to the overall communication traffic, there is a serious economic issue as faced by the service providers and is explained as follows. It is already ruled in countries like India that under the conditions of Internet neutrality, any VoIP application can be registered as an IP based service and can be used by the customers without incurring extra costs. Now, a

scenario can be considered where a consumer installs a VoIP based application (for example, Skype or WhatsApp) in the IP enabled mobile phone and uses that application to make voice calls instead of the cellular communication. This is both lucrative and cost-effective for the end-user as he has to pay only for the minimal data charges, instead of the cellular call charge which is usually higher, more so for international calls. On the other hand, this implies huge financial loss for the service provider whose main source of revenue is from the cellular communication. Therefore, one way to recover this loss is to lease a portion of the licensed spectrum using the Dynamic Spectrum Leasing (DSL) concept of CRN and allow external users (registered to other domains) to opportunistically access this licensed spectrum for VoIP and other applications.

4. ***Multiple Device Compatibility Scenarios:*** Another preferable use is to enable VoIP communication across different device categories including mobile phones, Wi-Fi enabled laptops, desktop computers and IP based portable devices. These devices may operate under different domains but can be integrated into using the CR platform for facilitating communication across different categories of users. Also, another application of this technology is the implementation of cost-effective solutions for educational and non-profit organizations by incorporating low-cost VoIP calls in licensed frequency bands after exploiting the concept of CR Networks.

5. ***Upcoming 5G Networks:*** Exponential growth of wireless traffic has led to the formation of the 5G network standard that is promised to satisfy multiple objectives in terms of data rate, latency, cost, energy and spectral efficiency, number of connected devices, etc. Implementing CRN is highly relevant in such scenarios, where novel concepts such as licensed spectrum sharing and dynamic spectrum leasing can be put to good use towards enabling seamless VoIP experience across diverse network categories [2.1, 2.50].

Despite all these advantages, VoIP communication by unlicensed users can result in severe degradation of the call quality unless adequate QoS strategies are defined to guide VoIP traffic in such active networks. A VoIP call has stringent requirements with respect to delay, jitter, packet loss and MOS that must be supported using a real-time QoS framework in the CRN, and is a subject of active research in the recent years [2.9, 2.51, 2.52]. Capacity analysis [2.9] followed by successive research on the joint packet-level and connection-level studies have already established the significance of hosting VoIP applications in CRN [2.51, 2.52].

However, there is a big difference between "analytical inferences" and "actual implementation" and herein lies the true challenge of this thesis work which is to implement the VoIP based CR systems over various platforms.

2.5 Mathematical Modeling of VoIP Traffic over a Wireless Channel in CRN

This section provides the basic overview of the analytical framework [2.9] comprising of PUs and VoIP based SUs in the CRN. This will help in determining the characteristics of both VoIP traffic and wireless channels in CRN and also enable us to determine important system metrics such as average number of active SUs, their mean arrival rate, transition probabilities for the channel states changing from Occupied to Unoccupied and vice-versa, etc.

Initially, the SU VoIP traffic is modeled as a simple on-off model. Let P_{on} be the probability that the SU is transmitting VoIP packets. Likewise, let P_{off} be the probability denoting the inactive status of the SU. The corresponding busy and idle times are exponentially distributed, whose mean values are given by $1/\alpha$ and $1/\beta$ respectively. Therefore, P_{on} and P_{off} are expressed as follows.

$$P_{on} = \frac{1/\alpha}{1/\alpha + 1/\beta} \quad (2.1); \qquad P_{off} = \frac{1/\beta}{1/\alpha + 1/\beta} \quad (2.2)$$

In addition, the VoIP traffic can be modeled using the two-state MMPP model [2.53] that captures the interframe dependencies between consecutive voice frames. Accordingly, the transition matrix (R) and the Poisson arrival rate matrix (Λ) of the MMPP model are expressed below.

$$R = \begin{bmatrix} -r_1 & r_1 \\ r_2 & -r_2 \end{bmatrix} \quad (2.3); \qquad \Lambda = \begin{bmatrix} \lambda_1 & 0 \\ 0 & \lambda_2 \end{bmatrix} \quad (2.4);$$

As MMPP model depends on the four parameters (r_1, r_2, λ_1, λ_2), the next step is to match them with the parameters of the on-off model (that is, α and β). In this regard, the IDC matching technique is used that provides the highest accuracy with appropriate computational complexity in comparison to other matching techniques [2.54]. Therefore, the MMPP model parameters are derived based on IDC and are expressed as follows.

$$r_1 = \frac{2(\lambda_2 - \lambda_{avg})(\lambda_{avg} - \lambda_1)^2}{(\lambda_2 - \lambda_1)\lambda_{avg}(\mathrm{IDC}(\infty) - 1)} \quad (2.5)$$

$$r_2 = \frac{2(\lambda_2 - \lambda_{avg})^2(\lambda_{avg} - \lambda_1)}{(\lambda_2 - \lambda_1)\lambda_{avg}(\mathrm{IDC}(\infty) - 1)} \quad (2.6)$$

$$\lambda_1 = A \cdot \frac{\sum_{i=0}^{N_{act\_avg}} i \cdot \pi_i}{\sum_{j=0}^{N_{act\_avg}} \pi_j} \quad (2.7)$$

$$\lambda_2 = A \cdot \frac{\sum_{i=N_{act\_avg}+1}^{N} i \cdot \pi_i}{\sum_{j=N_{act\_avg}+1}^{N} \pi_j} \quad (2.8)$$

It must be noted that N is the total number of VoIP based SUs, T_{basic} is the frame duration for the voice codec and A is the emission rate for the SU in the on state and is given by $1/T_{basic}$. Based on these parameters, the following expressions are derived.

1. Average arrival rate: $\lambda_{avg} = N \times A \times P_{on}$ (2.9)

2 . Average number of active SUs: $N_{act\_avg} = N \times P_{on}$ (2.10)

3 . Steady state probability of 1-D Markov Chain having N independent on-off voice users:

$$\pi_i = {}_N C_i \times P_{on}^i \times (1 - P_{on})^{N-i} \tag{2.11}$$

Once, the VoIP traffic is modeled, the next step is to characterize the wireless channels that are licensed to PUs and are hence occasionally available for the SUs (considering overlay CRN [2.5]). Such a wireless channel is modeled as a two-state Markov process [2.55].

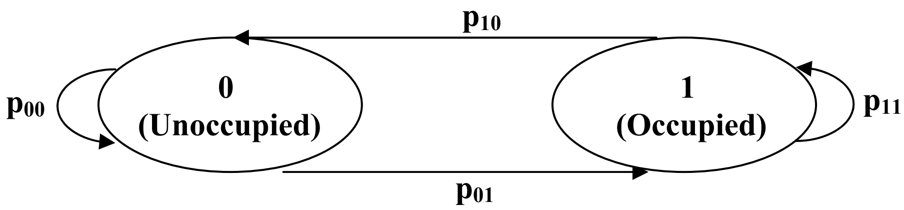


Fig. 2.16 Markov Model for a single wireless channel

As shown in Fig. 2.16, the "Occupied" state refers to a licensed channel not available to the SU as it is currently occupied by PU. "Unoccupied" state refers to an idle channel which can be utilized by the SU for transmissions. Considering M wireless channels, the transition probability ($P_{m,n}$) that there are m unoccupied channels (x_c) in the current frame and n unoccupied channels in the next frame is given by (2.12).

$$P_{m,n} = \sum_{x'=\max(0,m-n)}^{\min(m,M-n)} \binom{m}{x'} p_{01}^{x'} p_{00}^{m-x'} \binom{M-m}{y'} p_{10}^{y'} p_{11}^{M-m-y'} \tag{2.12}$$

where $y'=n-m+x'$ and x' and y' denote the number of channels whose status have changed from "Unoccupied" to "Occupied" and "Occupied" to "Unoccupied" states, respectively.

Thus, this section provides the basic mathematical formulation for VoIP transmission over a wireless channel in CRN. As novel design strategies are devised in the subsequent chapters, this analytical framework will be modified with the incorporation of relevant parameters and guided by necessary conditions.

2.6 QoS Metrics and Relevant System Parameters

This section provides a brief overview of the QoS metrics that characterize the VoIP sessions. Also, the relevant system parameters in CRN (that will be used for performance evaluation in subsequent chapters) are discussed in this section.

2.6.1 QoS Parameters for VoIP Applications

(i) Delay

VoIP delay or latency is defined as the total time required by speech to reach the listener's ear from the speaker's mouth. The permissible values of delay that can be tolerated by VoIP applications are standardized by ITU in Recommendation G.114 [2.56]. This recommendation defines three bands of one-way delay as shown in Table 2.3.

The various components of delay are further illustrated in Fig. 2.17.

Table 2.3 Delay Specifications for VoIP

Range in Milliseconds	Description
0-150	Acceptable for most user applications.
150-400	Acceptable provided that administrators are aware of the transmission time and the impact it has on the transmission quality of user applications.
Above 400	Unacceptable for general network planning purposes. However, it is recognized that in some exceptional cases, this limit is exceeded.

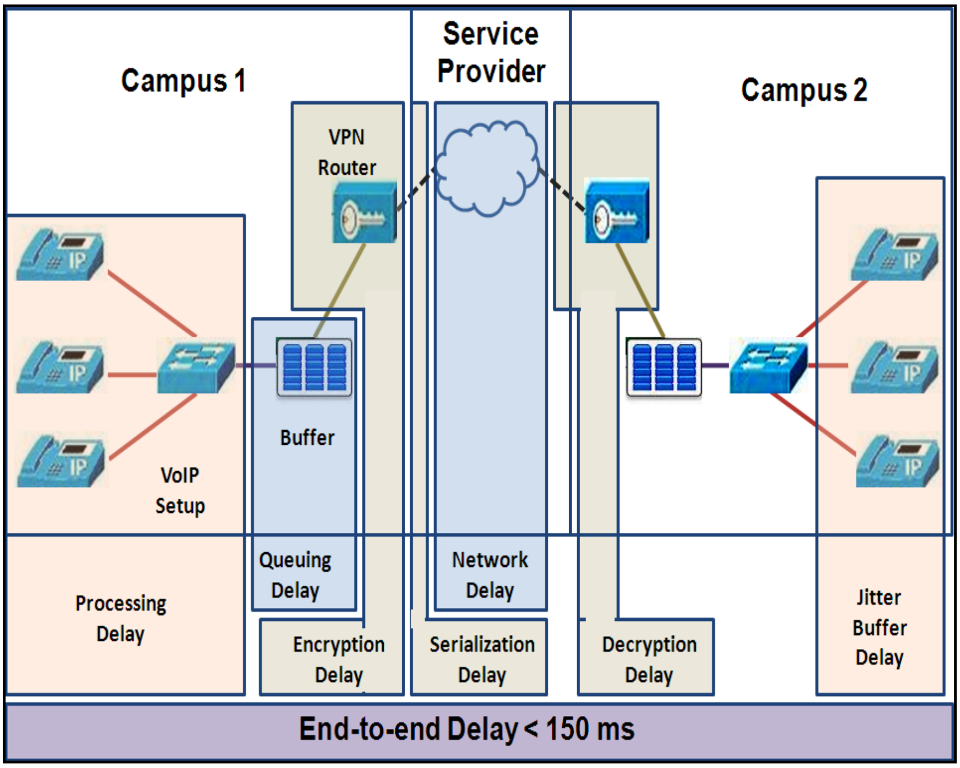


Fig. 2.17 Delay Components in VoIP System

(ii) *Jitter*

Jitter is the variation of packet inter-arrival time that exists only in packet-based networks. The difference between when the packet is expected and when it is actually received is jitter. Mathematically, jitter J can be represented by (2.13) and (2.14) [2.57].

$$D(i,j) = (R_j - R_i) - (S_j - S_i) = (R_j - S_j) - (R_i - S_i) \qquad (2.13)$$

$$J = J + (|D(i-1,i)| - J)/16 \qquad (2.14)$$

where S_i is the RTP timestamp from packet i, R_i is the time of arrival in RTP timestamp units for packet i and $D(i,j)$ is the delay for two packets i and j.

Jitter results due to out-of-order delivery of packets at the receiving end. While the sender in VoIP transmission is expected to reliably transmit voice packets at a regular interval based on codec characteristics, the packets may be delayed during transmission and arrive at different time periods, thereby resulting in jitter at the receiving end. Average jitter must be kept at less than 90 ms. While delay can disrupt the overall real-time communication, presence of considerable jitter degrades the receiving voice quality.

Jitter is minimized using jitter buffer that conceals the inter-arrival packet delay variation and transforms the variable delay into fixed delay component. Recommended practice is to count the number of packets that arrive late and create a ratio of these packets to the number of packets that are successfully processed. This ratio is used to configure the jitter buffer size and accordingly, the play out time for the initial packets in the queue is decided.

(iii) ***Packet Loss***

Packet loss is not tolerable in applications where user satisfaction is the primary constraint. VoIP is such an application where packet loss of more than 5% causes sufficient degradation in call quality. Packet loss is also used as a metric by many data protocols to know the condition of the network and thereby reduce the number of packets they are sending.

While voice is fed on data networks in the form of VoIP packets, it is essential to successfully transport voice in a reliable and timely manner. Moreover, it is useful to implement PLC [2.58] mechanisms to make the voice somewhat resistant to periodic packet loss. Cisco implements one such mechanism where whenever a voice packet is not received when expected, it is assumed to be lost and the last packet received is replayed. With G.729 [2.25] codec, it implies that the packet lost is only 20 ms of speech. Hence, the average listener does not notice the difference in voice quality.

(iv) ***Mean Opinion Score (MOS)***

Mean Opinion Score (MOS) gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using codecs [2.59]. It is calculated in two ways: subjectively and objectively. In the subjective voice testing, a group of listeners is given a sample of speech material and based on its quality, they give it a rating of 1 (bad) to 5 (excellent). The scores are then averaged to get the MOS [2.60]. On the other hand, the objective testing involves the calculation of MOS based on the values of R-Factor (discussed in the next section) and is given by the following expression.

$$
\begin{aligned}
MOS &= 1, \forall\ R \leq 0 \\
&= 1 + 0.035R + R(R-60)(100-R)7\times 10^{-6}, \forall 0 < R < 100 \\
&= 4.5, \forall\ R \geq 100 \ \textit{where}\ R = R_O - I_s - I_d - I_{e,eff} + A
\end{aligned}
\quad (2.15)
$$

MOS can also be obtained through implementation of software tools that carry out automated MOS testing in a VoIP deployment. Even though human perceptions are not taken into account in such scenarios, the advantage of having such tools is that they consider all the network dependency conditions that could influence voice quality. Some commonly used software include AppareNet Voice, Brix VoIP Measurement Suite, NetAlly, PsyVoIP, VQManager and VQmon/EP.

The permissible range of MOS values is listed in Table 2.4. A value of 4.0 to 4.5 is referred to as toll-quality and is an indication of a very high quality call.

Table 2.4 MOS Values and their Specifications

Values	Description
5	Perfect. Like face-to-face conversation or radio reception.
4	Fair. Imperfections can be perceived, but sound still clear. This is (supposedly) the range for cell phones.
3	Annoying.
2	Very annoying. Nearly impossible to communicate.
1	Impossible to communicate

Apart from evaluating the QoS of a particular VoIP call, MOS tests are also conducted to compare how well a particular codec works under varying circumstances that include different background noise levels, multiple encodes and decodes, and so on. Further, MOS can be used to compare between different VoIP services and providers.

(v) R-Factor

The R-factor uses a formula to take into account both user perceptions and the cumulative effect of equipment impairments to arrive at a numeric expression of voice quality. It is governed by the E-model as defined in ITU-T in Recommendation G.107 [2.61]. The E-model takes into account a wide range of telephony-band impairments, in particular the impairment due to low bit-rate coding devices and one-way delay, as well as the "classical" telephony impairments of loss, noise and echo and produces a scalar quality rating value known as "Transmission Rating Factor" or R-Factor.

The relation between the different impairment factors and R is given by the following equation [2.61].

$$R = R_O - I_s - I_d - I_{e,eff} + A \tag{2.16}$$

Here R_o is the basic signal-to-noise ratio, I_s represents all impairments that occur more or less simultaneously with the voice signal like too loud speech level, non-optimum sidetone, quantization noise, etc., I_d sums up all the impairments due to delay and echo effects, $I_{e,eff}$ is an "effective equipment impairment factor", which represents impairments caused by low bit-rate codecs, term A is an "advantage factor", which represents "advantage of access" for certain systems relative to conventional systems, trading voice quality for convenience. The range of R-Factor values with respect to VoIP transmission is enlisted in Table 2.5.

Table 2.5 R-Factor Values and their Significance

Range of E-model Rating R	User Satisfaction Level
90 > R < 100	Best - Very satisfied
80 > R < 90	High - Satisfied
70 > R < 80	Medium - Some users dissatisfied
60 > R < 70	Low - Many users dissatisfied
50 > R < 60	Poor - Nearly all users dissatisfied

2.6.2 System Parameters in CRN

(i) ***Spectral Efficiency:***

Since the licensed frequency spectrum allocated to the SUs is limited, it has to be utilized efficiently. A given bandwidth is used effectively only when maximum information can be transmitted over it. The term "Spectral Efficiency" is used to describe the rate of information being transmitted over a given bandwidth in specific communication systems. It can be calculated by dividing the total amount of data bits transmitted by the available bandwidth of the underlying channel. If a communication system uses one kilo hertz of bandwidth to transmit 1,000 bits per second, then it has a spectral efficiency or bandwidth efficiency of 1 (bit/s)/Hz.

Spectral efficiency assumes special significance for CRN [2.62] whose primary objective is to increase the overall spectrum utilization by allowing access to the idle spectrum bands. However, in an overlay CRN, there is a tradeoff between the level of protection offered to the PUs and the spectral efficiency achieved through incorporation of the SUs. Thus, the system capacity becomes another important issue in the context of spectral efficiency and is discussed next.

(ii) ***System Capacity:***

This is an important metric that indirectly implies the overall spectral efficiency of the CRN and can be defined as the maximum number of simultaneous SUs that can be admitted in a fixed frequency band. Unlike traditional networks, system capacity in CRN can vary depending on several factors including PU arrival rate, PU traffic distribution, variable channel conditions and heterogeneous application requirements of the SUs. In this regard, CRN must incorporate two important aspects; i) Call Admission Control (CAC) Policies and ii) Queuing Models. Some works [2.63] have also expressed the system capacity of CRN in the form of the SU Sum Goodput that is defined as the total amount of SU data successfully transmitted. It is to be noted that while increasing the system capacity can improve the spectral efficiency, it may degrade the energy efficiency of the system which is another important parameter and is described as follows.

(iii) Energy Efficiency:

Energy efficiency metric is an important one to be considered during network planning and is defined as the information bits transmitted per unit of transmitted energy. Considering the power constraints of mobile phones which are the primary users of VoIP applications, building energy efficient CRN [2.64, 2.65] is highly critical towards ensuring widespread acceptability. In such opportunistic communication models, energy can be lost while sensing as well as during transmission specifically in an overlay CRN. Accordingly, the CR timing cycle comprising of the sensing duration and the transmission duration plays a crucial role towards ensuring "green" communications and is discussed as follows.

(iv) CR timing cycle:

Every SU in a CRN is required to follow the CR timing cycle that comprises of sensing duration followed by the transmission interval. PU detection is performed in the sensing slot and if the channel is detected idle, transmission is resumed in the transmission duration. Obviously, there is a trade-off regarding optimal selection of these slots that is explained as follows. Increasing the sensing duration will provide higher protection to PU traffic as the probability of detecting PU presence will also increase. However, this will reduce the transmission time for the SU, thus resulting in loss of spectrum utilization [2.66]. On the other hand, increasing the transmission slot will ensure QoS guarantees for the VoIP calls by the SUs but will aggravate the risk of interference with the PU traffic. In general, the energy consumed as well as the throughput recorded through successful transmissions must be optimally maintained through proper configuration of these timing intervals.

(v) Spectrum Handoff Delay:

Spectrum mobility is a critical aspect of CRN that determines the long-term sustainability of the SU based applications in the system. One metric that quantifies this aspect is the spectrum handoff delay [2.5, 2.67]. This delay is unique to the CRN and is analogous to the handoff delay as observed in traditional wireless networks. Here, whenever PU arrives in the channel currently occupied by SU, the SU has to quit its transmission for the moment, select an idle channel and resume transmission in the newly selected channel after performing appropriate channel switching operations. This process is termed as spectrum handoff and the delay incurred therein is the spectrum handoff delay.

The spectrum handoff delay is the variable component delay and therefore adds to the jitter when dealing with real-time packet-switched traffic such as VoIP. It includes several delay components that are described as follows.

i) Channel switching delay: Actual switching of the channel requires shifting the operational frequency to that of the new channel and incurs a delay overhead on the part of the transceiver carrying out this task. This delay can be reduced through efficient hardware design of the radio terminal.

ii) Channel consensus delay: This delay follows the channel switching delay and is the time spent for achieving consensus in the new channel among the different SUs who are involved in communication. This consensus may be achieved through message passing and/or with the intervention of the SC node, and thus consumes a noticeable time overhead.

iii) Channel selection delay: Selecting an idle channel during handoff operation depends on the efficiency of the spectrum handoff policy adopted. The more the time spent in selecting this target channel, the higher the disruption in communication by the SU. This time spent is the channel selection delay. Reduction of this delay has been the primary focus of research in this domain of spectrum mobility. This is because this delay not only degrades the QoS for the underlying applications but also increases the probability of dropping from the channel (when the delay exceeds the threshold limit) which is not acceptable.

2.6.3 Quality of Experience

Quality of Experience (QoE) is a measure of the overall level of customer satisfaction with a particular vendor. This paradigm can be applied to any consumer-related business or service and is often used in information technology and consumer electronics. Although QoE and QoS are similar in their definitions, both are unique and should be separately considered during performance evaluation of the proposed system. QoE holds special relevance in CRN, where momentary disruptions in voice communication are common and may lead to increased volumes of customer dissatisfaction. This section provides a brief overview of this metric.

Precisely, QoE expresses the satisfaction of the end customer, both subjectively and objectively. It is therefore user-dependent and depends on the psychological aspects of the customers. Major factors affecting the QoE include cost, reliability, efficiency, privacy, security, interface, user-friendliness and user-confidence. At the same time, external factors include the user's terminal hardware (e.g. landline or cellular), the working environment (e.g. fixed or mobile) and the importance of the underlying applications (e.g. text or voice/video).

European Telecommunications Standards Institute (ETSI) has clearly defined the distinctions between the QoS and QoE with respect to wireless communication [2.68] and the same holds true for VoIP applications as well. Accordingly, QoE for real-time communication can be defined as the overall acceptability of an application or service, as perceived subjectively by the end-user. This definition has two significant aspects.

i) Encompasses the complete effects of the end-to-end system (including client, terminal, network, services, infrastructure, etc.)

ii) Acceptability in terms of user expectations and behavior

For the success of the communication purposes, the overall quality must be judged from three perspectives as stated below:

i) Technically centered Quality of Service (QoS)

ii) User perceived QoS (also known as QoP: Quality of Perception)

iii) Quality of Experience (QoE)

These three qualities are related to each other and must be tuned in accordance with the overall objective of increasing the “Quality” of the communication. The relation is shown in Fig. 2.18.

With respect to VoIP, QoE can be easily illustrated with an example. “If a phone call of 30 minutes duration falters at the last minute when most likely people exchange greetings, the overall QoE of the communication will fall rapidly, even though the QoS is maintained satisfactorily”. Although mostly subjective, the voice QoE for VoIP call can be technically expressed as [2.69].

Voice QoE = (Quality of the Network Delivery Stream + Quality of encoding components to the Network + Quality of decode from the network + Human Factors) per unit time. (2.17)

QoE assumes special significance in such opportunistic networks like CRN where there can be sudden and random disruptions in communication. Deploying effective strategies with respect to spectrum management and mobility may lead to better QoS values, but these do not always guarantee the overall QoE (as evident from the previous example). It is apparent that simulation and analytical studies may provide us with the parameters to define and evaluate the QoS for ongoing VoIP sessions. Whereas, QoE evaluation requires actual deployment in a practical test-bed to observe how a long duration VoIP call is sustained in the CRN despite occasional PU presence. This requires recording statistics for long duration calls under widely varied network conditions, and thereafter analyzing their variance with respect to the threshold values. This is crucial because even if the mean values for the QoS metrics remain within their threshold limits, QoE may be hampered if the recorded statistics show significant variance in their readings. Thus, apart from QoS, QoE must also be minimized to ensure the overall success of the QoS aware CR system for supporting VoIP communication.

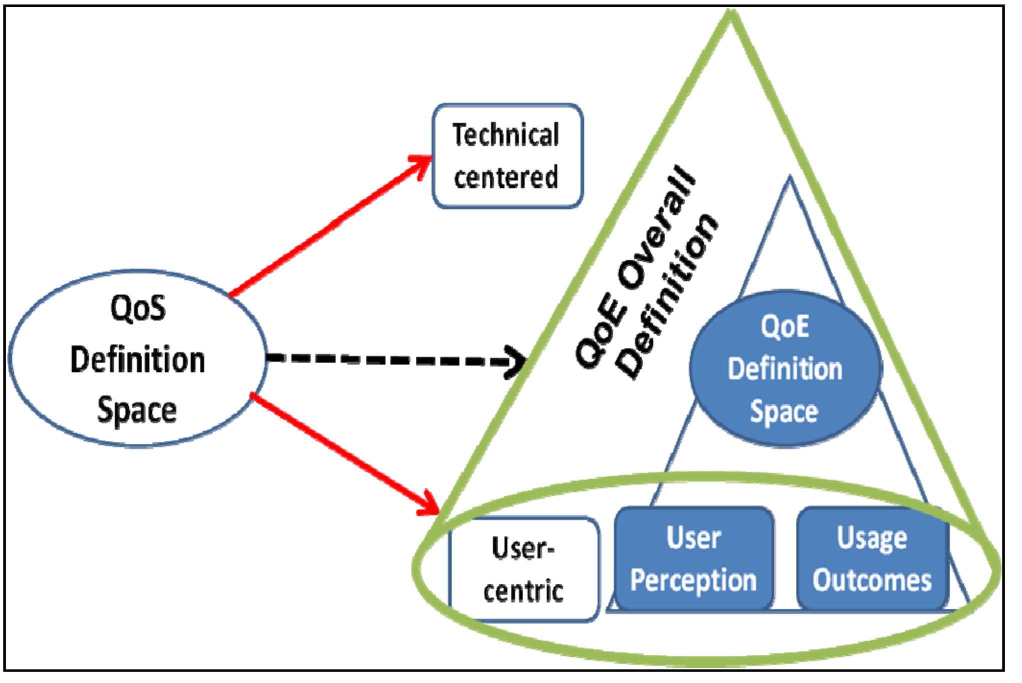
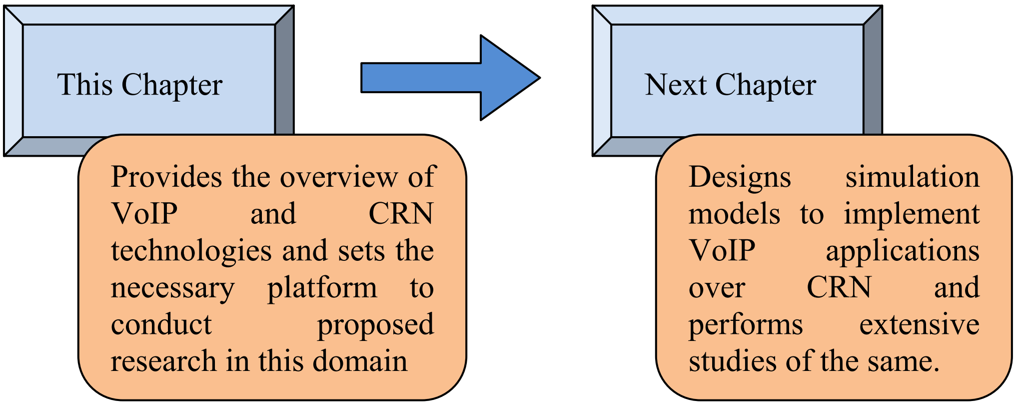


Fig. 2.18 Definition Domain for Quality of Experience (QoE) [2.68]

2.7 Summary

This chapter has laid the foundation for the proposed research study in this thesis by highlighting the relevance of implementing VoIP applications over CRN in the context of modern-day communications. In this regard, a brief overview of these technologies is provided including their evolution and practical significance. The basic mathematical model related to VoIP traffic and wireless channel in CRN is also presented. The VoIP traffic is modeled as a simple on-off traffic model and mapped to the MMPP process to capture the inter-frame dependencies. The wireless channel on the other hand is modeled as a two-state Markov chain and its transition probabilities are calculated from Occupied to Unoccupied states and vice-versa.

Next, the important parameters are discussed with respect to VoIP QoS metrics (delay, jitter, packet loss, MOS and R-Factor) and CRN based system parameters (energy efficiency, spectral efficiency, system capacity, spectrum handoff, etc.) followed by a brief description of the QoE metric. It is subsequently inferred that the appropriate tuning of CRN parameters will inadvertently result in enhancement of QoS for VoIP transmissions, however at the cost of degraded values for some system metrics. Thus, there must be a trade-off in terms of improving system parameters in CRN while still maintaining the call quality at acceptable limits. This trade-off is achieved in the subsequent chapters through the formulation of different design policies followed by suitable implementations in simulation and test-bed models.



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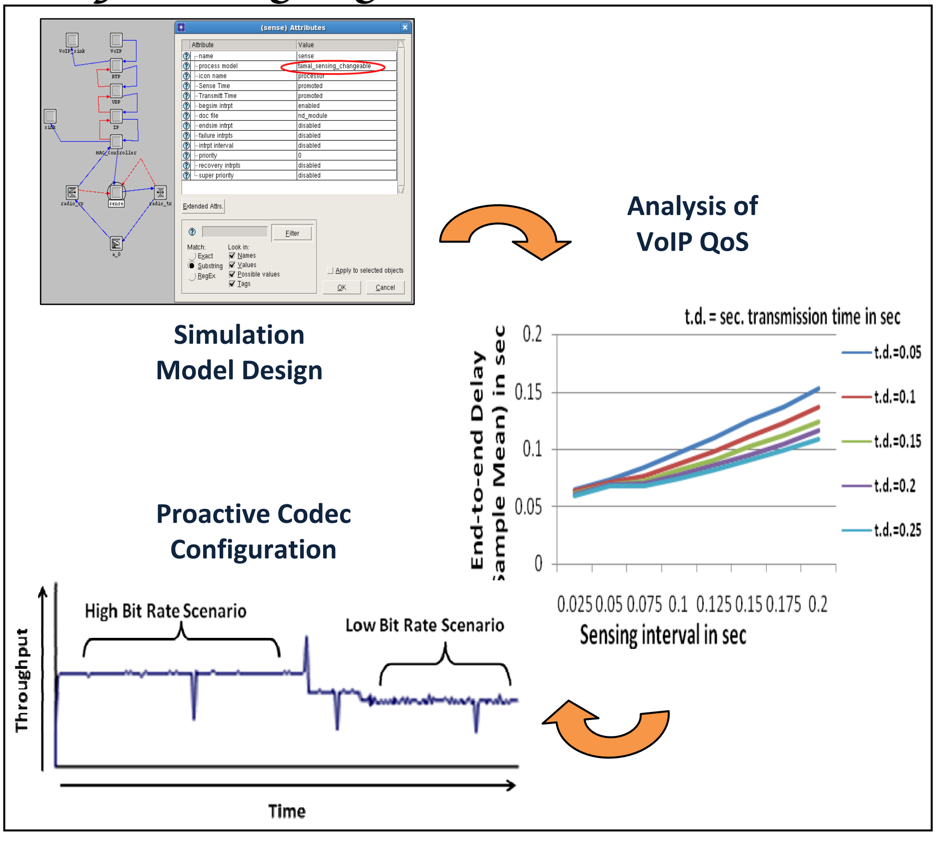
Chapter 3.

DESIGN OF SIMULATION MODELS FOR QOS AWARE VOIP COMMUNICATION OVER CRN

Chapter Highlights



CHAPTER 3: Design of Simulation Models for QoS Aware VoIP Communication over CRN

"Design is not just what it looks like and feels like. Design is how it works"

- Steve Jobs, Apple Inc.

Outline of the Chapter

As clearly evident from the previous chapters, stringent QoS requirements in VoIP coupled with complexities in CRN have initiated intensive research in this emerging discipline with respect to performance analysis and optimizations that are suitably guided by simulation results. Accuracy and credibility of the simulation output in these studies are strongly dependent on the proper design of the simulation model that must have a strong mathematical foundation. ***However, in the absence of any such standard model in literature for VoIP applications over CRN, the first step towards conducting full-fledged research in this domain is to develop a comprehensive simulation model with sufficient credibility***. This model will serve as a platform to conduct in-depth analysis of existing policies in VoIP and CR domain that will subsequently necessitate the formulation of design methodologies in this thesis.

Accordingly, the objective of this chapter is to build standard models for SUs in CRN and successfully implement VoIP applications in those models, which will serve as the initial point for development with respect to all future simulation studies in the category of VoIP over CRN. Initially, the models of VoIP based SUs in CRN are developed using OPNET Modeler 16.0.A following distributed architecture in single-channel and multi-channel scenarios and further in Visual C++ adhering to the principles of centralized architecture. These models are validated by the comparison of simulation results obtained in both these platforms. The underlying mathematical model behind the proposed design is established and the critical factors pertaining to both VoIP and CRN domain are extensively analyzed.

In the next phase, based on the outcome of the simulation analysis, ***a proactive codec adaptation algorithm is proposed involving VoIP codecs and active queues to ensure high throughput of VoIP traffic with enhanced QoS***. Implementation of the algorithm in the designed simulation model witnesses significant performance improvement in VoIP call quality with reduced packet loss ratio and increased throughput during ongoing VoIP transmissions by SUs.

3.1 Introduction

In absence of any established standard for VoIP implementation over CRN, research in this realm is primarily based on simulation studies and mathematical analysis. However, model description of VoIP over CRN in the literature lacks transparency and clarity and also requires proper documentation that must be equipped with suitable verification and validation. It has generally been observed that simulation studies in telecommunication and networking often fall short of credibility due to inappropriate use of pseudo-random generators and lack of an in-depth analysis after conducting the simulations [3.1]. Therefore, the design of a proper simulation model is as crucial as the analysis and optimization of challenges involved in providing QoS aware VoIP transmissions over CRN. Additionally, the simulation output from the designed model must be analyzed using mathematical tools and validated accordingly so as to facilitate their utilization in future research activities globally [3.2].

This chapter therefore aims to develop models of VoIP in CRN in different simulation platforms to help researchers with further exploration in this realm. Keeping pace with rapid and continuous development in this domain, several modifications are to be made in the proposed models with the focus on applying novel, adaptive and cross-layer strategies. Moreover, these models should be modified with suitable mathematical framework in every network layer to support advanced opportunistic mode of communication in CRN. Performance in different types of networks can be further studied by placing such customized nodes in already established network models like WSN, WBAN, etc. On top of that, proper configuration of VoIP agents is a necessity for successful VoIP deployment in CRN and this includes careful monitoring of codec parameters with implementation of adaptive optimization policies.

In this regard, the motivation behind our work is established following an exhaustive literature survey in the following section.

3.1.1 Literature Survey

This chapter deals with two significant aspects namely, i) simulation model design and analysis; and ii) codec adaptation for QoS aware VoIP communication over CRN. Accordingly, relevant research studies in these areas are investigated that will set the foundation for the proposed design work in this chapter.

(i) ***Literature Survey: Simulation Model Design***

Considering the fact that the first notable work in the domain of VoIP and CRN was done in the year 2010 by H. Lee et. al. [3.3], only limited research activity as in this work has indeed ventured into the aspect of simulation model design and analysis for executing VoIP applications over CRN. This model design was carried out in 2011 (only a year after the feasibility of VoIP over CRN was first established in 2010). It is therefore imperative that very few models actually did exist at that point of time that provided researchers with a platform to implement their customized protocols relating to VoIP and CR domain.

Nonetheless, the literature survey in this section is conducted based on the overall research findings in the broad domain of simulation model design in CRN. To cite them, firstly a cross-layer design approach is applied in [3.4] with the formulation of Distributed Opportunistic Spectrum Access (D-OSA) scheme in MATLAB where multiple CR users attempt to access the licensed spectrum bands of the PUs. Another notable work in [3.5] has used MATLAB and Simulink and performed hardware co-simulation with Xilinx System Generator to model the different multi-modulation schemes for baseband processing section of SDR. Likewise, OMNeT++ is used as the simulation tool to model different aspects of CRN IN [3.6]. Other studies using OMNeT++ include the design of CR Mac Protocol in [3.7], iterative water-filling techniques to model the sub-channel concept in [3.8], and the design of CR Mac Protocol with an array of static and re-configurable radios to develop CR-NIC Module in [3.9].

Using existing programming languages, models as in [3.10] are designed in C++, where the authors have implemented a modular CR node with typical TCP/IP protocol stack. Another model in [3.11] also builds infrastructure-based CRN platform using C++. In addition, three notable works are reported in literature [3.12], which specifically use NS2 [3.13] for simulation modeling. Firstly, CRCN (Cognitive Radio Cognitive Network) in [3.14] executes dynamic spectrum resource allocation along with power management and routing protocols for CR users. The second work implements an adhoc-based CRN in CRAHN (Cognitive Radio Adhoc Network) [3.15] where overlay based CR users perform opportunistic channel access by periodically sensing the channels. Finally, in the third model, the MAC and Physical layer issues are addressed through system modeling in CogNS [3.16].

It is thus evident from the literature survey that real-time QoS issues for VoIP users have not been considered in detail during model design in simulation platforms, and this forms the primary motivation behind the proposed work in this chapter.

(ii) __Literature Survey: Codec Adaptation Algorithms__

Every codec adaptation algorithm follows a decision policy, which dictates the usage of voice codecs and the manner in which this adaptation is performed [3.17]. Non-adaptive policies, for example, prevent the calls from any adaptation of codec bit rates as described in [3.18]. Single adaptive policies, on the other hand, allow changes in codec bit rates for only those calls that are responsible for variation in link capacity as followed in [3.19]. Multi adaptive policies, as witnessed in [3.20], are current areas of active research and allow calls to adapt its codec properties based on link conditions and related factors.

Various works as suggested in [3.21] and [3.22] have focused on the switching of codecs such that more number of calls can be accepted at a tolerable limit. Evaluation of congestion due to rate change is analyzed in [3.19] and codec change is proposed for the node suffering rate change. In [3.23], when the channel is detected congested (based on Real-Time Control Protocol (RTCP) packet loss and delay feedback information), a central element performs codec adaptation using transcoding methods for calls entering wireless cell. Focus shifts from multi-rate effects to congestion due to existence of additional VoIP sessions in [3.24] where the algorithm highlights on adapting low priority calls in favour of high priority ones by “down-switching” the codec and packetization interval of the low priority calls. Codec adaptation strategies can also be formulated according to the arrival of new calls or during rate changes or in case of both these scenarios [3.17].

However, these works focus on “reactively” varying the codec properties, that may result in subsequent throughput degradation. On the other hand, proactive codec adaptation is yet to be performed in these studies. Further, only few works, such as the one in [3.25] have ventured into the domain of centralized codec management where the codec properties are tuned in accordance with a centralized entity (for example, the Access Points or APs). Also, no such work has been reported in literature, where the codec properties are proactively configured for sustaining long duration VoIP communication over CR networks.

Therefore, all these shortcomings are addressed through the design of a centralized proactive codec adaptation algorithm in this chapter that is implemented in the designed simulation models and evaluated for performance efficiency in terms of maintaining sustainable VoIP sessions in the CRN.

3.1.2 Significant Contributions

The major contributions of this chapter are summarized as follows.

1. Extensive studies of the current research activities in literature in Section 3.1 reveal lack of a comprehensive model for implementing VoIP applications over CRN. Accordingly, the objective of this work is clearly highlighted in Section 3.1.

2. The problem definition behind the model design is discussed in Section 3.2 and the basic approach towards designing the CR Model is highlighted.

3. A model for VoIP over CRN is designed in OPNET Modeler 16.0.A for both single and multiple channel scenarios based on the distributed architecture in Section 3.3. This is followed by the detailed analysis of VoIP performance under the basic CR scenarios with respect to delay, jitter, packet loss, throughput, etc.

4. Extended analysis of the critical attributes in the developed models is carried out in Section 3.4. This involves the study of simulation outcome under several factors including SU traffic distribution patterns, different PU activity models, PU detection issues and different packet processing delays in the SU node.

5. Section 3.5 provides another model design of VoIP based CRN by adhering to the principles of centralized architecture. Performance analysis is carried out with respect to throughput and packet loss during VoIP transmissions by the SUs.

6. The underlying mathematical model behind the simulation design is established in Section 3.6. Expressions for PU detection probability in the

sensing as well as transmission intervals, along with the total CR cycle time as available to the SUs are derived.

7. Thereafter, a proactive codec adaptation algorithm is proposed in Section 3.7 involving VoIP codecs and RED buffers. The algorithm is implemented and analyzed in the designed models with respect to total obtained throughput and packet loss ratio for the admitted SUs.
8. Finally, the advantages of the proposed models along with the areas of further improvement are discussed in Section 3.8. Validation of the model design is also performed by comparison of simulation output from developed models in order to ensure the credibility of the obtained results and the inferences derived henceforth.

Finally, the chapter is concluded in Section 3.9.

To the best of our knowledge, no such work covering all these aspects related to CRN based model design in simulation followed by extensive analysis and codec adaptation operations specifically related to VoIP traffic has been reported in literature so far, and this establishes the novelty of the work in this chapter.

3.2 Problem Definition

The objective of this chapter is to develop efficient models of VoIP over CRN in different simulation platforms that will help in performing analytical studies and implementing proposed optimizations by suitable modifications to the proposed models. Adhering to the principles of Design Science methodology [3.26], the primary aim of this work is the analysis of problem domain followed by development and evaluation of simulation models to study the behavior of real-time VoIP transmission in CRN.

The driving factor behind implementation of VoIP over CRN stems from the fact that link utilization and hence, capacity of VoIP system must be increased along with overall enhancement in the call quality [3.3]. Let L be the link utilization with respect to time and M be the derived expression from Mean Opinion Score (MOS). Let $f(L,M)$ be the function of L and M. Therefore, the objective is to maximize *f(L,M)*. Hence, the objective function can be mathematically denoted by (3.1).

$$T = max\left[f\left(L, M\right)\right] \tag{3.1}$$

Active networks like VoIP transmission in CRN embed computational capabilities into conventional networks, thereby massively increasing the complexity. Therefore, simulation has scored over traditional analytical methods in analyzing such active networks [3.27]. This necessitates appropriate design of models in the simulation platforms based on an underlying principle. In this regard, the basic principle behind proposed simulation model is shown in Fig. 3.1.

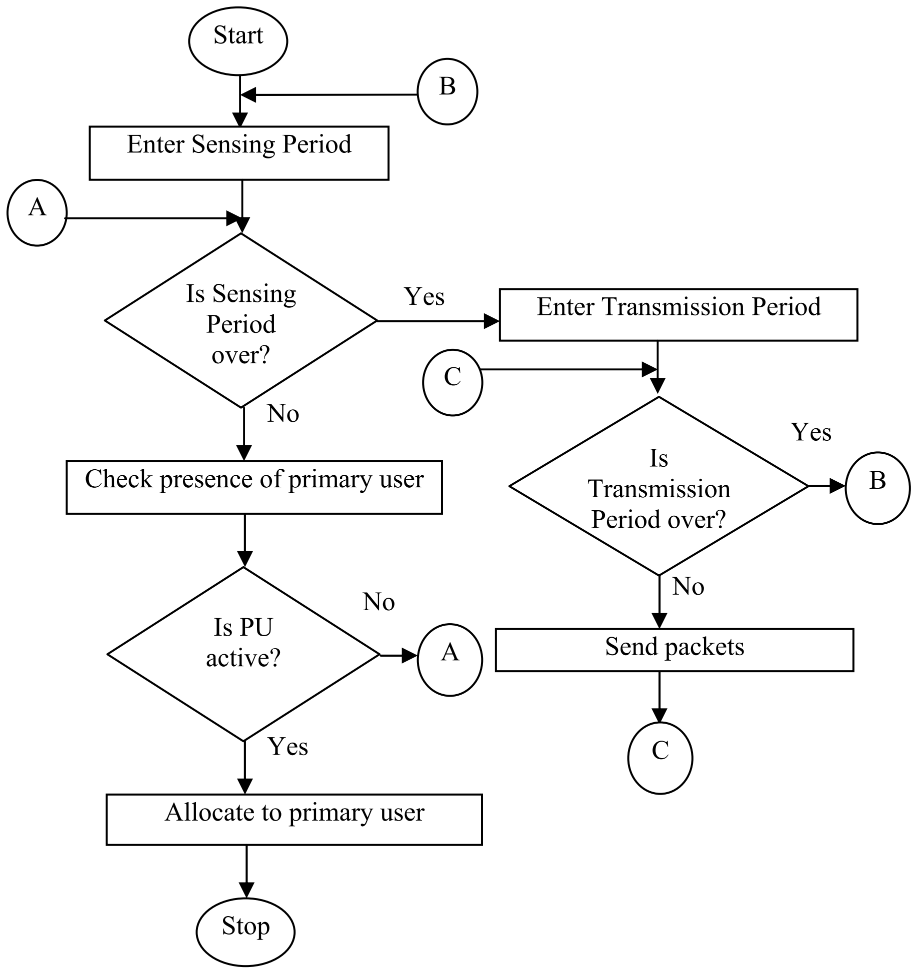


Fig. 3.1 Flowchart depicting the proposed CR approach where SUs sense the channel for PU presence before transmission

PU generates traffic at uniform distribution interval. The SU is the VoIP user and implements G.711 codec [3.28]. The PU is the licensed user with priority to use the channel. But it does not always occupy channels and SUs as CR users are permitted to use these channels in absence of PU. An SU senses the assigned channel in the sensing period and starts transmission in the transmission period only when the PU is inactive.

3.3 Design and Analysis of Simulation Model in OPNET Modeler 16.0.A

Based on the design principle, a simple model for VoIP over CRN is initially designed in OPNET Modeler 16.0.A [3.29] following distributed architecture [3.30]. OPNET Modeler 16.0.A is chosen as the simulation platform as it offers highly customizable nodes along with options ranging from traffic distribution and network parameters to cross-layer architecture based operational modes and collection of wide range of statistical results.

At first, an overview of the simulator is discussed followed by detailed design and analysis of VoIP communication over CRN.

3.3.1 Overview of OPNET Modeler 16.0.A

OPNET Modeler [3.29] provides a comprehensive development environment supporting the modeling of communication networks and distributed systems. Both the behavior and performance of modeled systems can be analyzed by performing discrete event simulations. The OPNET Modeler environment incorporates tools for all phases of a study, including model design, simulation, data collection, and data analysis.

OPNET Modeler users can be divided into two broad categories namely,

1. **System modelers**: System Modelers are the traditional users of OPNET Modeler, who study technical problems using simulation. They are usually interested in performance measures and behavioral analysis of a proposed system. Often they are the designers of networks, network devices and protocols, or distributed applications.

2. **Authors**: Authors do not use OPNET Modeler to conduct their own simulation studies, but rather to prepare an environment for others to do

so. OPNET Modeler is used to create customized models that are correct for a particular type of end user.

(i) ***OPNET Modeler Architecture***

OPNET Modeler 16.0.A provides a comprehensive development environment for modeling and performance evaluation of communication networks and distributed systems. The package consists of a number of tools, each one focusing on the certain aspects of the modeling task. These tools fall into three major categories that correspond to the three phases of modeling and simulation projects, namely i) Model Specification and Modeling Communications with Packets, ii) Data Collection and Simulation, and iii) Analysis.

These phases are performed in sequence. They form a cycle, with a return to Specification following Analysis. Specification is divided into two parts: initial specification and re-specification, with only the latter belonging to the cycle. The simulation phases are illustrated in Fig. 3.2.

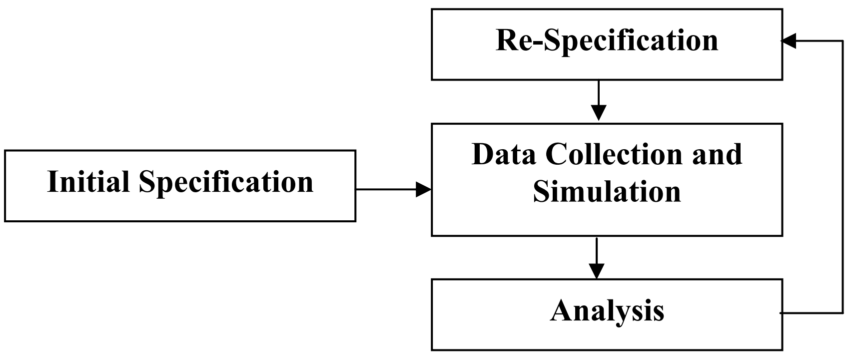


Fig. 3.2 Simulation Phases of OPNET Modeler 16.0.A

OPNET Modeler 16.0.A makes use of graphical specification of models wherever appropriate. Therefore, the model-specification editors present a graphical interface in which the user manipulates objects representing the model components and structure. Each editor has its own specific set of objects and operations that are correct for the modeling task on which it is focused. The primary editors are shown in Fig. 3.3. Some of the most essential editors are listed below.

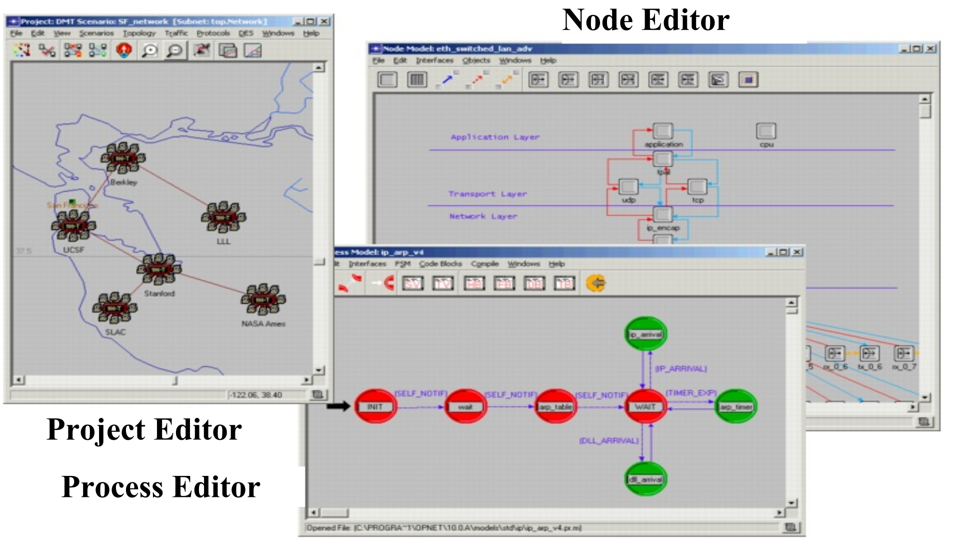


Fig. 3.3 Primary editors of OPNET Modeler 16.0.A

1. **Project Editor:** Develop network models. Network models are made up of subnets and node models. This editor also includes basic simulation and analysis capabilities.

2. **Node Editor:** Develop node models. Node models are objects in a network model. Node models are made up of modules with process models. Modules may also include parameter models.

3. **Process Editor:** Develop process models. Process models control module behavior and may reference parameter models.

4. **External System Editor:** Develop external system definitions. External system definitions are necessary for co-simulation.

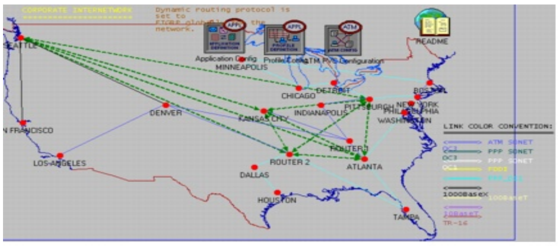
5. **Link Model Editor:** Create, edit, and view link models.

6. **Packet Format Editor:** Develop packet formats models. Packet formats dictate the structure and order of information stored in a packet.

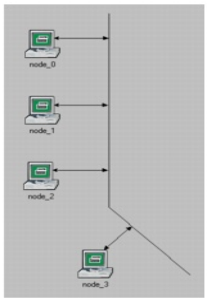
(ii) ***<u>Modeling Domains</u>***

The Network, Node and Process models are referred to as the modeling domains of OPNET Modeler 16.0.A because they span all the hierarchical levels of a model. These are described as follows.

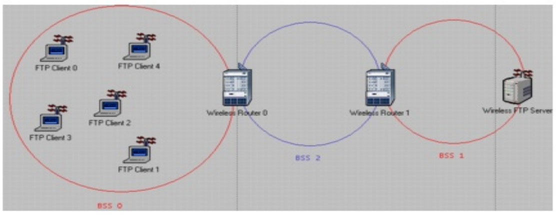
1 . **Network Domain:** The Network Domain's role is to define the topology of a communication network. The communicating entities are called nodes and the specific capabilities of each node are defined by designating their model. The Project Editor can provide a geographic context for network model development. In addition to providing an intuitive environment for deploying the components of a network model, this feature provides an intrinsic notion of distance, which allows automatic calculation of communication delays between nodes. The screenshot is provided in Fig. 3.4.



Point-to-Point Links



Bus Links



Radio Links

Fig. 3.4 Network Domain of OPNET Modeler 16.0.A

2 . **Node Domain:** The Node Domain performs modeling of communication devices that can be deployed and interconnected at the network level. In OPNET Modeler terms, these devices are called nodes, and in the real world, they may correspond to various types of computing and communicating equipments such as routers, bridges, workstations, terminals, mainframe computers, file servers, fast packet switches, satellites, and so on. Node models are developed in the Node Editor as depicted in Fig. 3.5 and are expressed in terms of smaller building blocks called modules.

3 . **Process Domain:** Processes in OPNET Modeler 16.0.A are based on process models that are defined in the Process Editor. A screenshot of Process Node in OPNET Modeler 16.0.A is shown in Fig. 3.6.

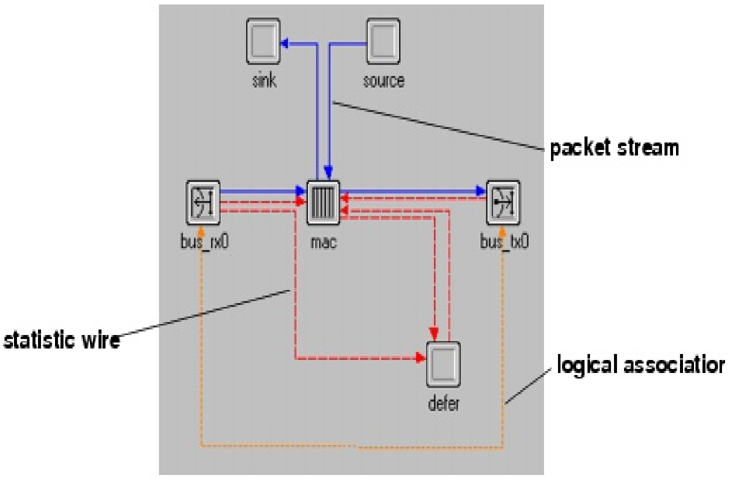


Fig. 3.5 Node Domain of OPNET Modeler 16.0.A

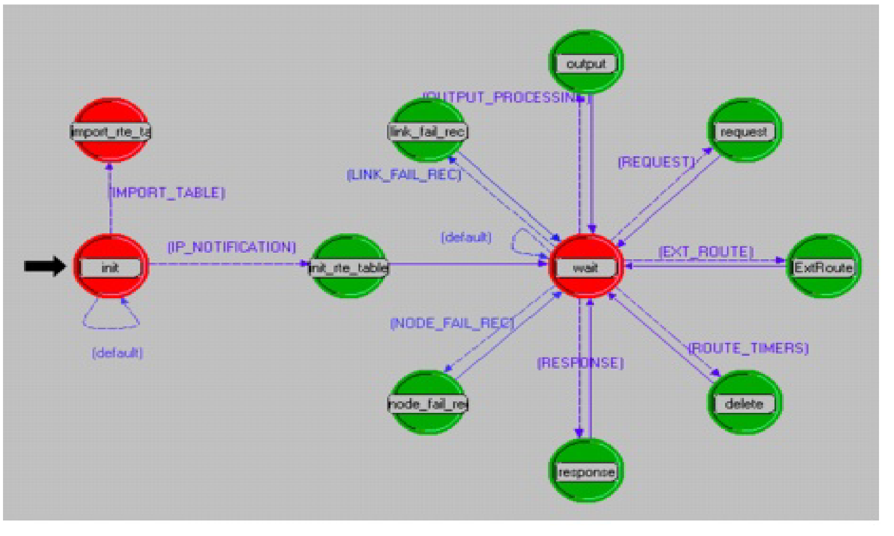


Fig. 3.6 Process Domain of OPNET Modeler 16.0.A

The relationship between a process model and a process is similar to the relationship between a program and a particular session of that program running as a task (in fact, the term "process" is used in many operating systems as well). Just as nodes created in the Project Editor are instances of node models defined with the Node Editor, each process that executes in a queue, processor, or e.sys module is an instance of a particular process model.

The Process Editor expresses process models in a language called Proto-C, which is specifically designed to support development of protocols and algorithms. Proto-C is based on a combination of State Transition Diagrams (STDs), a library of high-level commands known as Kernel Procedures, and the general facilities of the C or C++ programming language.

(iii) Typical Applications of OPNET Modeler16.0.A

OPNET Modeler 16.0.A can be used as a platform to develop models of a wide range of systems. Some examples of possible applications are listed below with specific mention of supporting features:

1. **Standards-based LAN and WAN performance modeling:** Detailed library models provide major LAN and WAN (Wide Area Network) protocols. Configurable application models are also provided by the library, or new ones can be created.

2. **Internetwork planning:** Hierarchical topology definitions allow arbitrarily deep nesting of sub-networks and nodes and large networks are efficiently modeled. Scalable, stochastic, and/or deterministic models can be used to generate network traffic.

3. **Research and development in communication architectures and protocols:** OPNET Modeler 16.0.A allows specification of fully general logic and provides extensive support for communication related applications. Finite state machines provide a natural representation for protocols.

4. **Distributed sensor and control networks, on-board systems:** OPNET Modeler 16.0.A allows development of sophisticated, adaptive, application-level models, as well as underlying communication protocols and links. Customized performance metrics can be computed and recorded, scripted and/or stochastic inputs can be used to drive the simulation model, and processes can dynamically monitor the state of objects in the system via formal interfaces provided by the statistic wires.

5. **Resource sizing:** Accurate, detailed modeling of a resource's request-processing policies is required to provide precise estimates of its performance when subjected to peak demand (for example, a packet switch's processing delay can depend on the specific contents and type of each packet as well as its order of arrival). Queuing capabilities of

Proto-C provide easy-to-use commands for modeling sophisticated queuing and service policies.

6. **Mobile packet radio networks:** Specific support for mobile nodes, including predefined or adaptive trajectories, predefined and fully customizable radio link models, and geographical context based models are provided by OPNET Modeler network specification environment.

7. **Satellite networks:** Specific support for satellite nodes, including automatic placement on specified orbits, a utility program for orbit generation, and orbit visualization and orbital-configuration animation program are given for customization by the OPNET Model library.

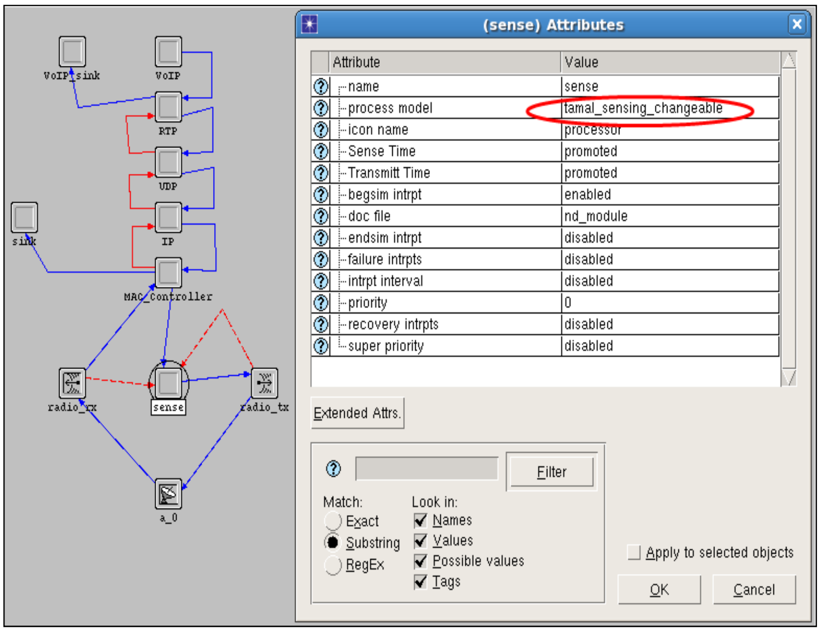
8. **C3I and tactical networks:** OPNET Modeler also supports diverse link technologies, modeling of adaptive protocols and algorithms in Proto-C, notification of network component outages and recoveries, scripted and/or stochastic modeling of threats, determination of friendly interference and jamming, etc.

3.3.2 Design in a Single-Channel Scenario

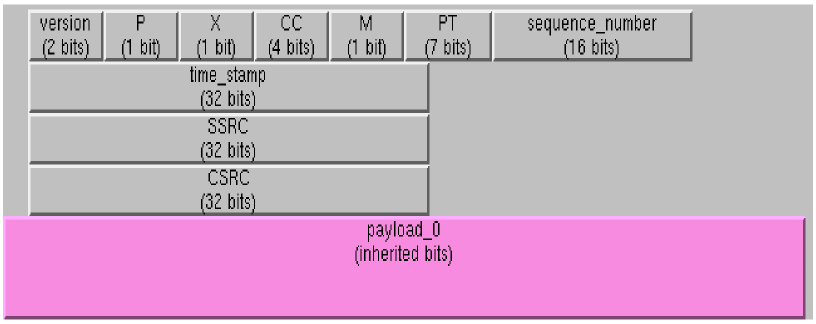
Relying on the benefits and efficiency of the OPNET simulator, a model for VoIP in CRN is designed over OPNET Modeler 16.0.A in this section with respect to a single channel scenario following the principles of distributed architecture.

(i) ***Design Overview***

The node model for SU as designed in OPNET Modeler 16.0.A is shown in Fig. 3.7 (a). The application layer node is the VoIP node responsible for managing VoIP transmissions. This node is followed by Real-Time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP) nodes. The functionalities of network layers are incorporated in process models corresponding to each node in the node model. *VoIP_sink* node acts as sink for packets already received and processed accordingly. The *MAC_Controller* node acts as link layer node and cooperates with physical layer node which is involved in sensing, transmission and reception. Spectrum management functionalities like spectrum sensing and spectrum handoff must work in collaboration with the communication protocols [3.30] and hence such cross-layer architecture is implemented in this chapter.



(a)



(b)

Fig. 3.7 (a) Node Model of VoIP over CRN for single channel scenario, and (b) RTP Packet Format in OPNET Modeler 16.0.A

The process model corresponding to *MAC_Controller* node is highlighted in the *Attributes* Tab of Fig. 3.7 (a). It consists of sense and transmit processes that respectively sense and transmit packets according to the principle as stated in Section 3.2. Single radio architecture is implemented in the sensing principle, where a specific time slot is allocated for spectrum sensing. Thus, only certain accuracy can be guaranteed for spectrum sensing results. It also results in a decrease in spectrum efficiency as some portion of available time slot is used for sensing instead of data transmission [3.31]. However, the advantage of single radio architecture is its simplicity and lower cost [3.32], both of which are essential for low cost VoIP communication. Moreover, sensing PU presence is based on energy detection based radiometry [3.33] or periodogram. The advantages include low computational and implementation complexities that make the model simpler with less algorithmic delays. It is also generic as receivers need no knowledge on PU's signal [3.32]. However, it may lead to false alarms [3.34] and precautions must be taken to eliminate them. Finally, VoIP packets are designed following the protocol formats in each layer. For example, the designed RTP packet is illustrated in Fig. 3.7 (b).

(ii) ***Discussion of Simulation Results***

The simulation model is analyzed to evaluate the essential QoS metrics that define the overall call quality in VoIP communication. Many simulation runs are conducted to study the impact of the basic CR cycle parameters (sense time, transmission time) during the ongoing VoIP sessions. Accordingly, the sensing and transmission time for SUs are varied.

As observed from Fig. 3.8, mean end-to-end delay for VoIP calls in SUs rises with increase in sensing period due to increase in waiting time for packets waiting to be transmitted. This delay is decreased with rise in secondary transmission period, thereby creating a favorable environment for VoIP transmission. It is further witnessed from Fig. 3.9 that higher sensing periods increase SU packet loss which further rises with a decrease in the secondary transmission period.

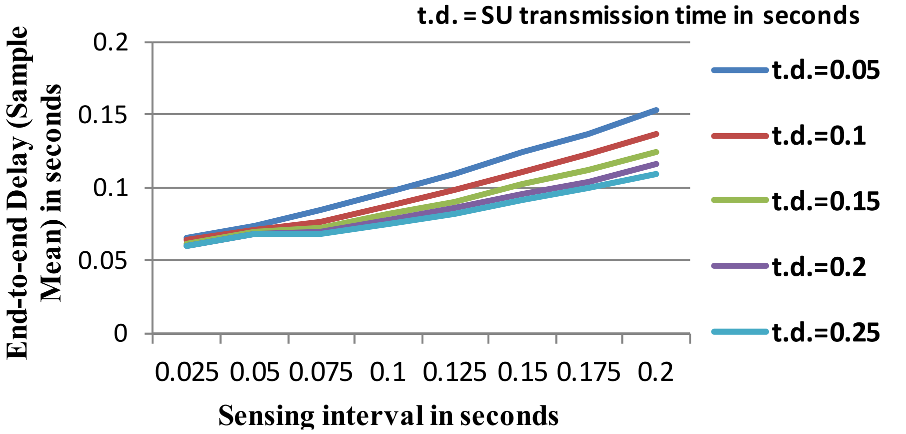


Fig. 3.8 Variation in end-to-end delay (sample mean) with sensing and transmission intervals of SUs

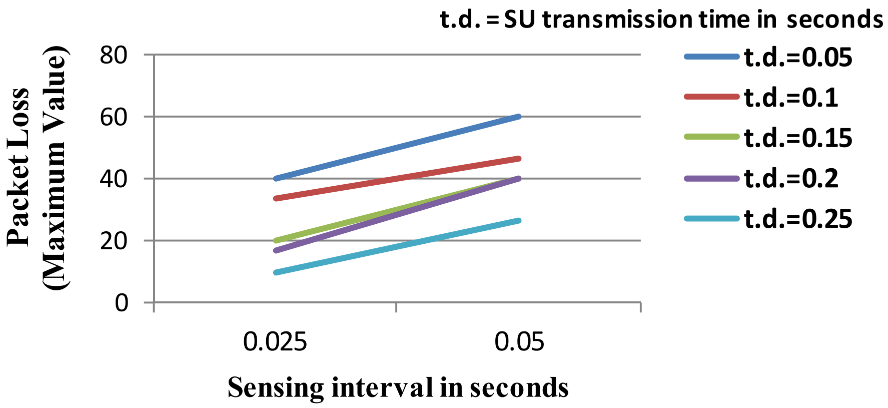


Fig. 3.9 Variation in packet loss (maximum value) with sensing and transmission intervals of SUs

Throughput degradation for SUs is recorded in Fig. 3.10 with increase in sensing period. A sharp decline in received total traffic with higher values of sensing intervals denotes that there must be a maximum bound on the sensing duration for successful VoIP transmission. Finally, it is observed from Fig. 3.11 that jitter rises with increase in sensing period and decrease in secondary transmission period.

Thus the study of simulation results makes it clear that sensing and transmission intervals have profound effect on the QoS of the VoIP calls. It is evident that sensing time must have both upper and lower bounds to allow successful VoIP communication without interfering with PU traffic. A solution to this problem includes increasing the sensing period resulting in low SU throughput. Another alternative solution is to use short sensing and transmission cycles which further increase jitter in voice traffic. Therefore, proper configuration of sensing and transmission durations is a tricky problem with respect to the SUs, especially when dealing with the real-time QoS requirements of VoIP traffic.

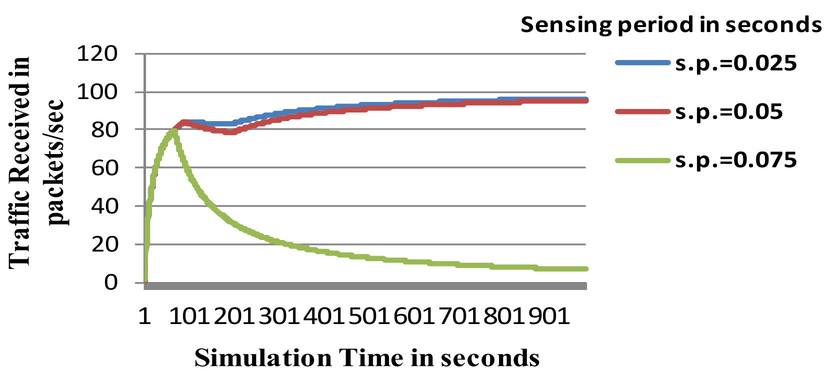


Fig. 3.10 Variation in traffic received for varying sensing intervals of SUs

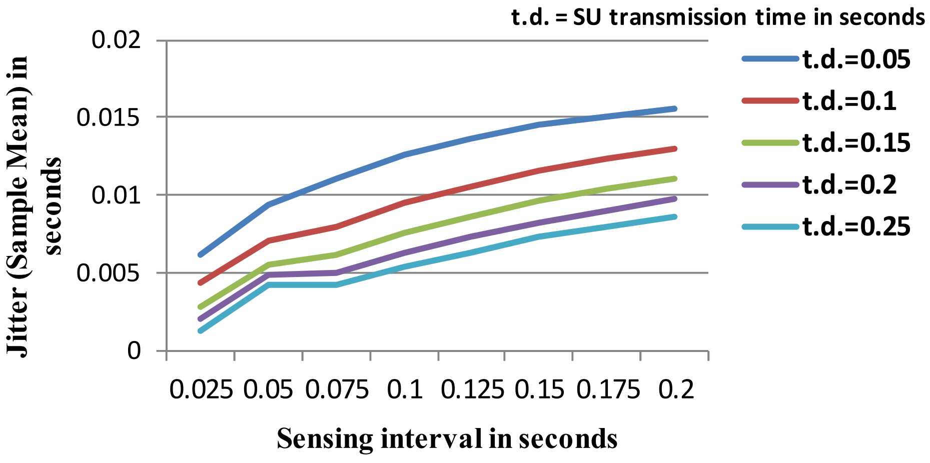


Fig. 3.11 Variation in jitter (sample mean) with sensing and transmission intervals of SUs

3.3.3 Design in a Multiple-Channel Scenario

The work in the previous section is now extended in a multiple-channel scenario with the assumption that at any point of time, atleast one channel is free for transmission for certain time duration.

(iii) <u>Model Overview</u>

The modified node model for SU as designed in OPNET Modeler 16.0.A is shown in Fig.3.12. The node model as described in Section 3.3.2 is updated to include another set of transmitter and receiver that operates in a different channel. This necessitates design of two sense nodes for two channels. Therefore, *MAC_Controller* node is developed accordingly to switch to the free channel when the other channel is busy. The other nodes along with packet formats are kept similar to the ones described in Section 3.3.2. Single-radio architecture and radiometry as discussed in Section 3.3.2 are applied to this model.

(iv) <u>Study of Simulation Output</u>

The developed model is again used to analyze the VoIP performance in CRN following the principle as stated in Section 3.2. The channel throughput is shown in Fig. 3.13 for each channel from the sender end.

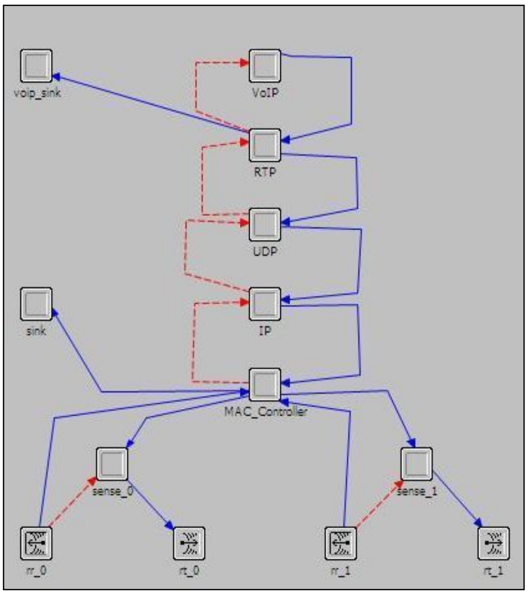


Fig. 3.12 Node Model of VoIP over CRN in OPNET Modeler 16.0.A for multiple-channel scenario

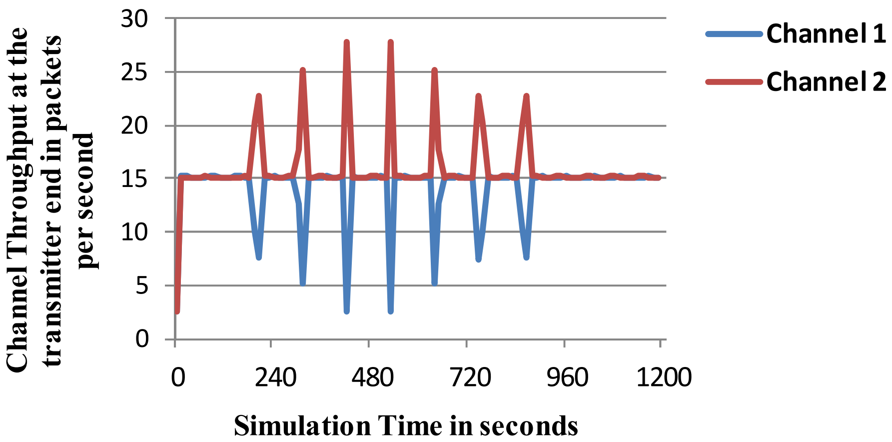


Fig. 3.13 Variation in channel throughput at the transmitter for 2 channels in CRN during ongoing VoIP session

Sense and transmit durations are kept at 1 sec. It is observed that in the presence of PU, channel 2 initiates VoIP sessions and hence simultaneous transmission is observed in both the channels.

Fig. 3.14 records the time average view of channel throughput at the receiver end. It is noted that as simulation time progresses, throughput from channel 1 decreases with rise in throughput in channel 2 due to high PU activity.

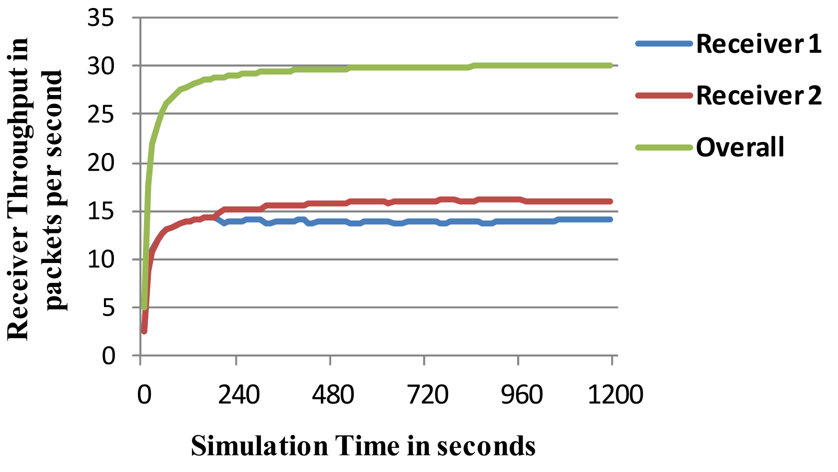


Fig. 3.14 Variation in throughput (time average) at the receiver for 2 channels in CRN during ongoing VoIP session

However, multi-channel setup ensures successful packet reception from both channels as indicated from the overall throughput of the receiver. For different sensing and transmission intervals, it is observed from Fig. 3.15 that channel throughput increases with increasing transmission duration.

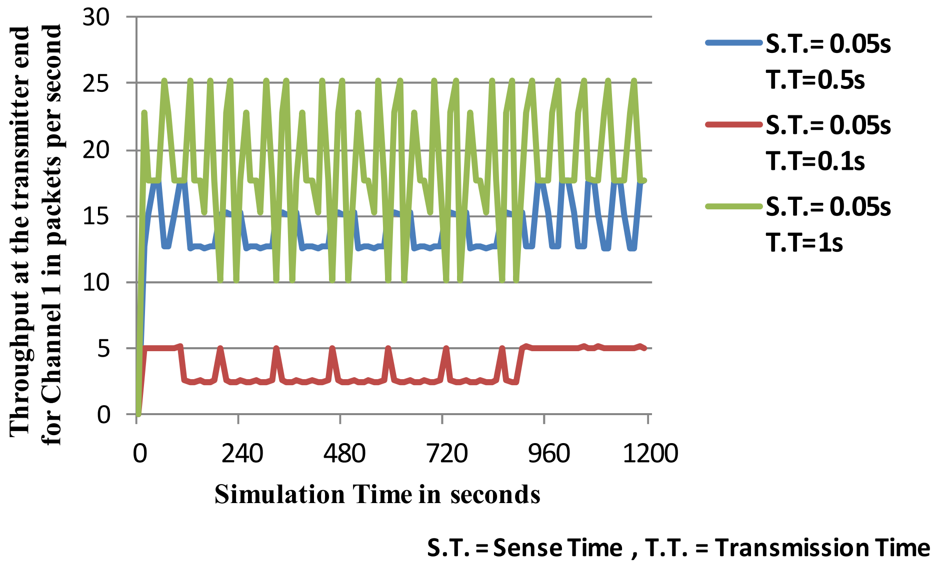
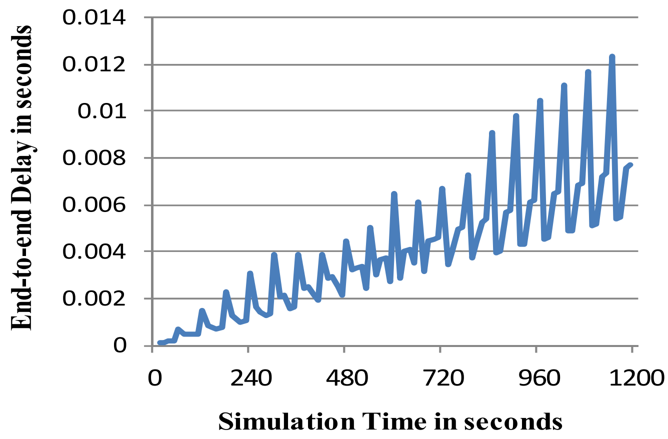
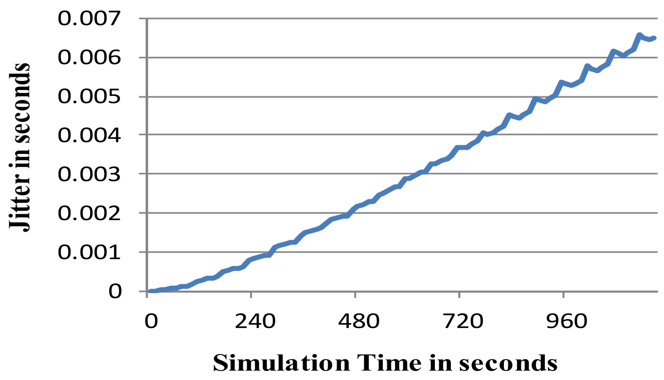


Fig. 3.15 Variation in channel throughput at transmitter end for channel 1 with sensing and transmission intervals of SUs for ongoing VoIP session

Finally, it is clearly observed in Fig. 3.16 (a) and Fig. 3.16 (b) that the end-to-end delay and jitter increase with simulation time as the call progresses.



(a)



(b)

Fig. 3.16 Variation in (a) end-to-end delay, and (b) jitter during ongoing VoIP session

This is because the number of channel switching occurrences increases with increased PU activity leading to higher number of spectrum handoffs [3.35]. Hence, both delay and jitter increase under such circumstances.

Therefore, the multi-channel scenario provides us with an outlook on how the spectrum handoff and load balancing occur when SUs switch their transmissions from one channel to the other. This assumes special significance in CRN, where PUs can arrive in the SU occupied channels at any time leading to disruptions in VoIP calls by SUs and increase in interference for the PUs.

It is to be noted that due to simulation constraints, gathering important statistics in the designed models becomes a challenging issue once and after the simulation duration exceeds a certain threshold limit. Hence, in those cases, the timing intervals have been suitably modified to collect and study the data. This is because the primary focus here has been laid towards observing the trends rather than absolute values.

3.4 Analysis of Critical Attributes in the Designed Models

After the preliminary study of QoS metrics for VoIP calls over CRN, the critical factors affecting the basic model design are discussed in this section, thoroughly. Analysis is carried out based on the obtained simulation data from the OPNET Modeler 16.0.A based Model.

3.4.1 SU Traffic Distribution Pattern

The proposed design considers deploying VoIP applications by the SUs. Hence, the traffic distribution for SUs is a major factor towards model design and development. It is worth mentioning that VoIP transmission occurs in talkspurts with multiple on-off periods [3.36]. Therefore, the silence periods in VoIP transmission render the channel idle and also reduce the probability of sudden interference with PU. This fact is clearly established in Fig. 3.17 which shows the variation in channel throughput for varying on and off periods.

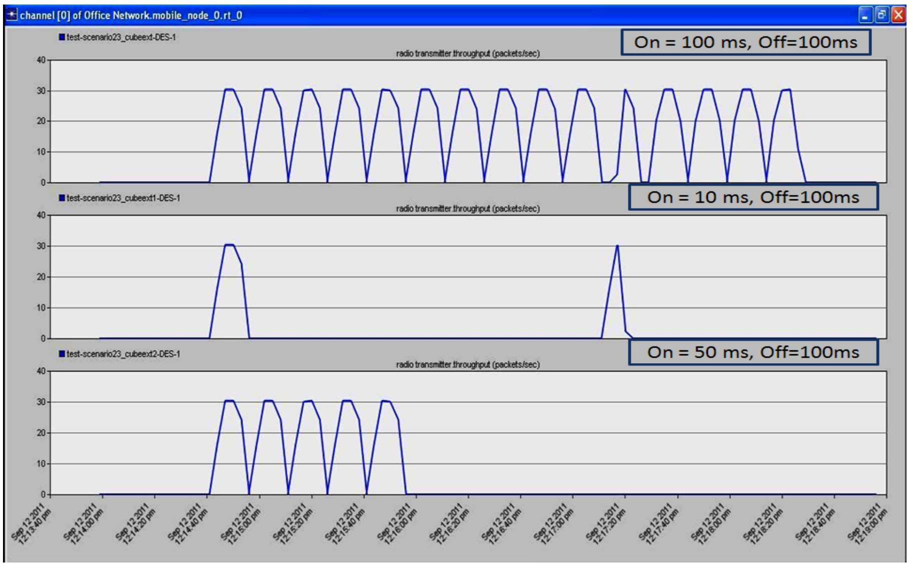


Fig. 3.17 Variation in SU throughput for different on-off periods

Moreover, the variation in SU transmission duration further affects the VoIP talkspurts and the throughput varies accordingly as reflected from Fig. 3.18. However, there is a clear trade-off between increasing SU transmission duration (and subsequent increase in VoIP call quality) and minimizing the probability of PU interference.

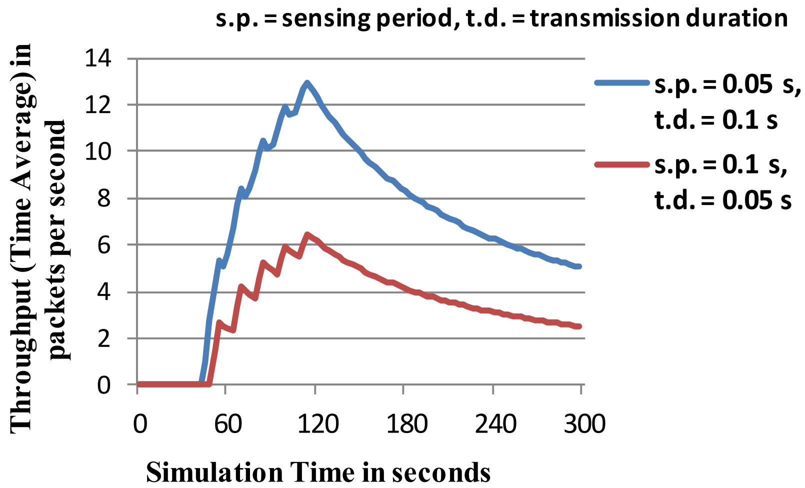


Fig. 3.18 Variation in SU talkspurt based throughput for different sensing and transmission durations of SUs

Finally, SU traffic distribution is varied in Fig. 3.19 that results in different levels of throughput obtained in case of VoIP SUs for identical transmission durations and on-off periods. Therefore, it can be inferred that an accurate estimation of SU traffic pattern is crucial towards achieving high throughput for VoIP call while maintaining an acceptable call quality.

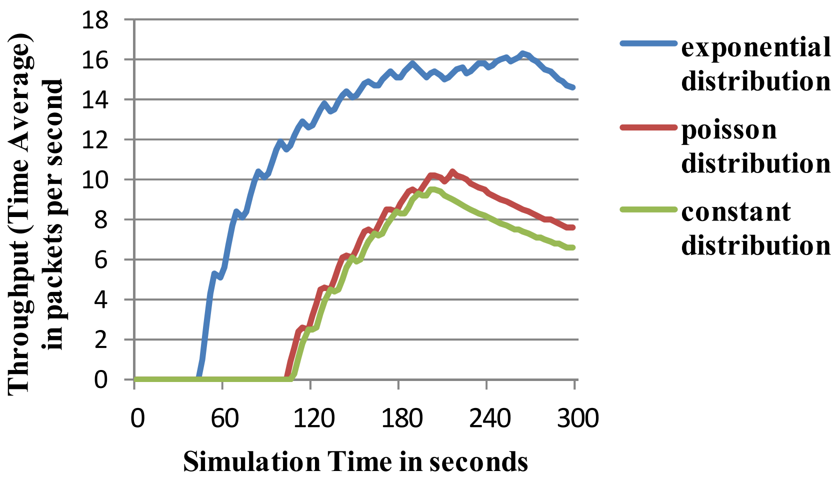


Fig. 3.19 Variation in SU talkspurt based throughput for different traffic distributions of SUs

3.4.2 PU Activity

SU activity is further influenced by sudden variations in PU traffic distributions. PUs being the licensed users must be protected from any possible interference with SUs [3.37]. Therefore, detailed analysis of PU traffic distribution on VoIP throughput is carried out in the developed OPNET based model. It is observed from Fig. 3.20 that increased PU presence results in decrease in SU throughput on the current channel along with subsequent rise in SU transmission on the alternative channel. This fact reflects the necessity of carefully selecting alternative channels for SU transmission in conjunction with channel reservation policies [3.38].

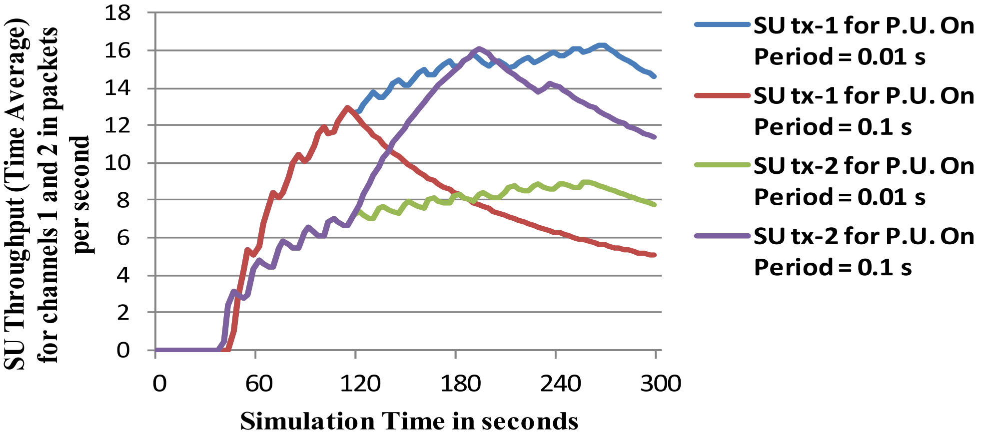


Fig. 3.20 Variation in SU talkspurt based throughput for different PU traffic activities

It is also observed from Fig. 3.21 that modifications in sensing and transmission durations along with variation in PU transmission rate jointly affect the rate of interference between the SU and PU transmissions. This is denoted in the following figure by the instances of collision between PU and SU transmissions.

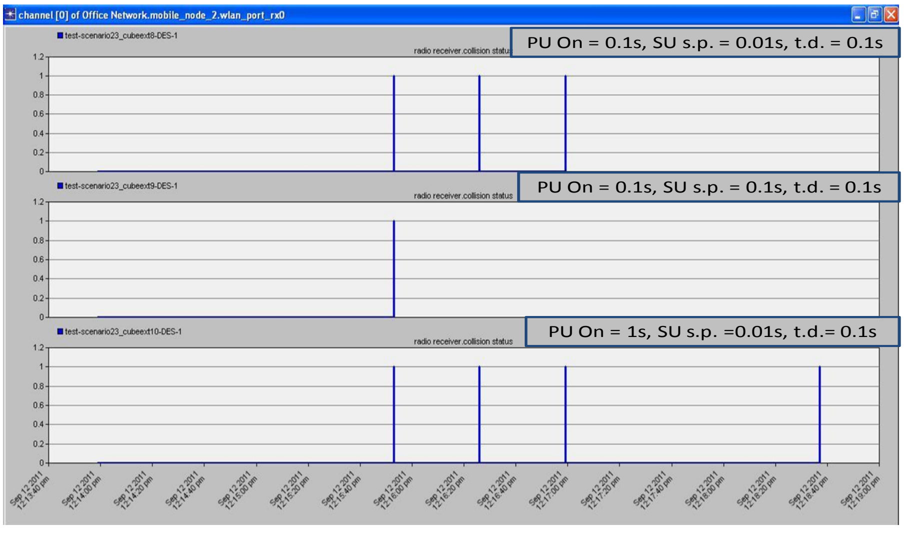


Fig. 3.21 Interference between transmissions of SU and PU for varying PU activities and SU parameters

Finally, the PU traffic distribution pattern is varied to analyze the effect on SU throughput. Fig. 3.22 clearly points out that SU load balancing on multiple channels is a critical aspect that must be studied extensively for different PU transmission distribution functions.

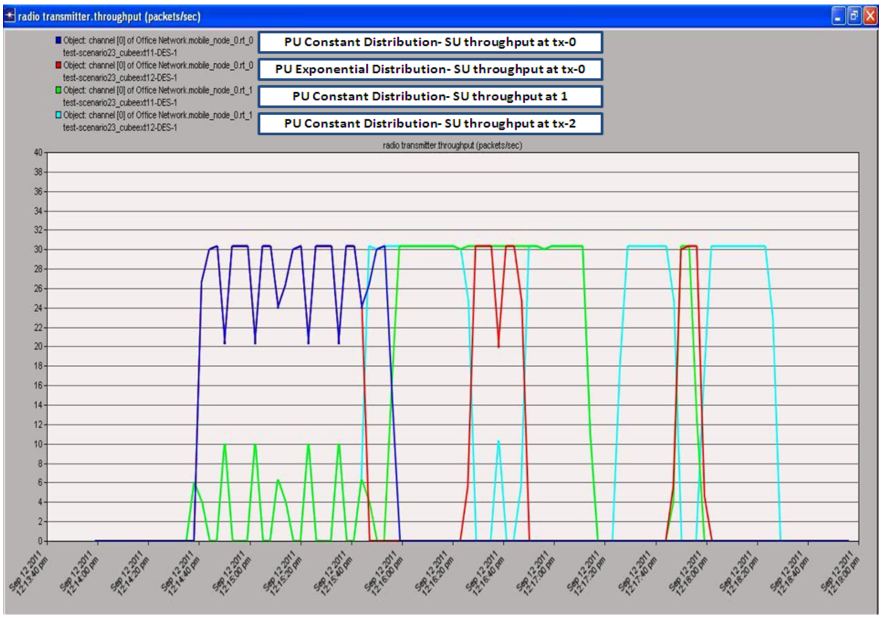


Fig. 3.22 SU throughput in both channels for varying PU traffic distributions

3.4.3 PU Sensing Issues

Detailed analysis of the effect of PU presence on SU throughput has already been performed in the previous sections. Hence, it is obvious that appropriate PU detection technique must be deployed by SU to avoid any interference and subsequent degradation in VoIP call quality. The reason for applying periodogram technique in this context has been mentioned in Section 3.3.2. However, delays involved in the detection of PU presence and subsequent decision-making operations can considerably decrease the efficiency of periodogram, resulting in perceivable interference [3.30]. This fact is established in Fig. 3.23 where higher delays in sensing PU result in increased duration for which the active channel is marked as idle even though it is actually busy, creating a scenario of miss-detection [3.30]. Subsequently, SU transmission time increases, thereby increasing the collision with PUs. Therefore, such delays are highly critical towards designing a comprehensive simulation model and must be reduced.

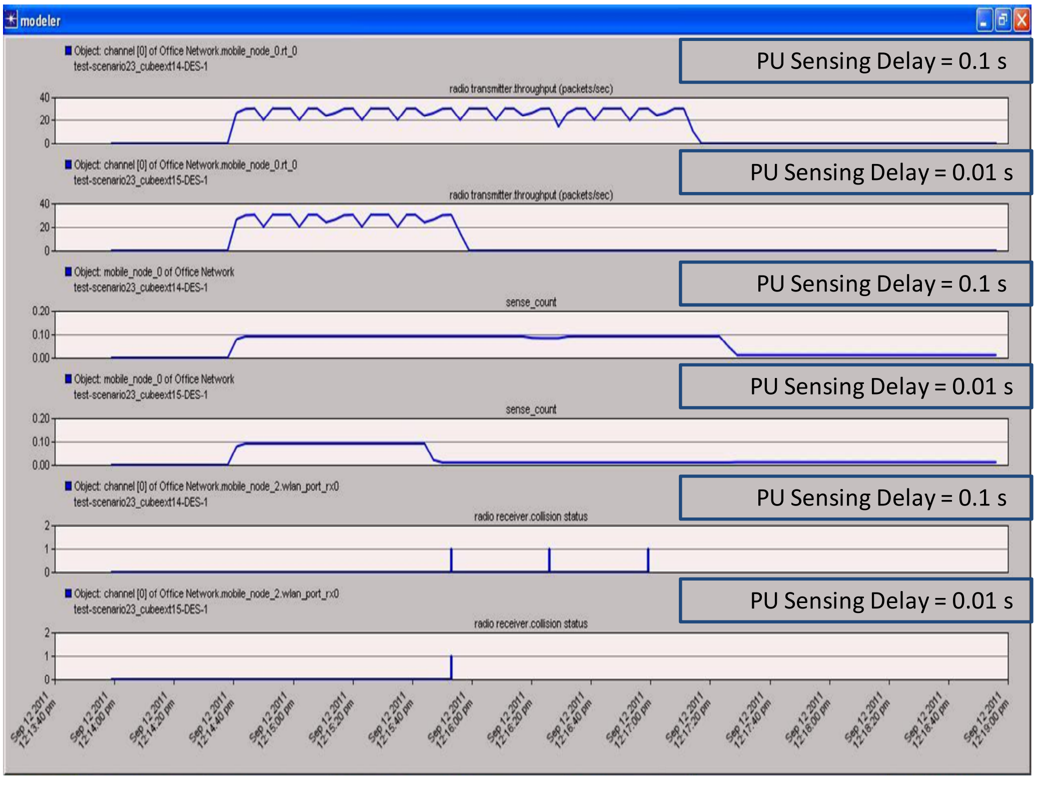


Fig. 3.23 Variation in SU transmission duration, time for which the channel is marked as "idle" and collision status with PU for varying delays in sensing PU presence

3.4.4 Packet Processing Delays

Any packet-switched network is characterized by delays involving packet generation, packet processing and reception [3.39] and this model is not an exception. Such delays adversely affect the overall quality of transmission and can prove to be critical for real-time VoIP applications in intelligent networks like CRN. This model therefore captures such delays and indicates the corresponding degradation suffered by VoIP applications. It is observed from Fig. 3.24 that a rise in the delay in the arrival of VoIP packets for transmission results in increased collision with the PUs. This is because SU transmission does not occur on its designated time slot. Packet generation delay further induces loss in throughput at the transmitter and receiver ends as observed in Fig. 3.25 (a) and Fig. 3.25 (b) respectively.

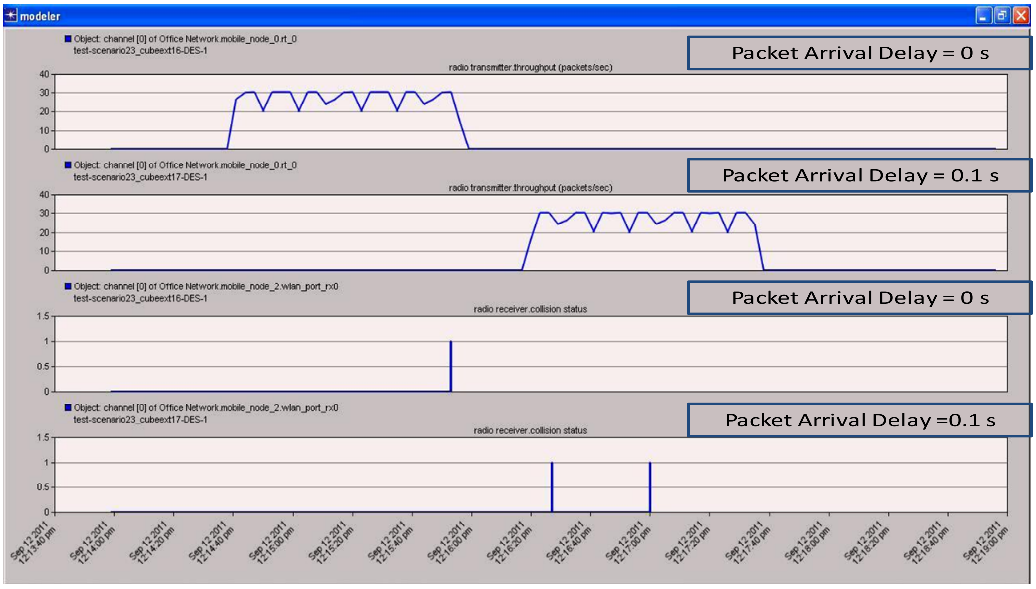
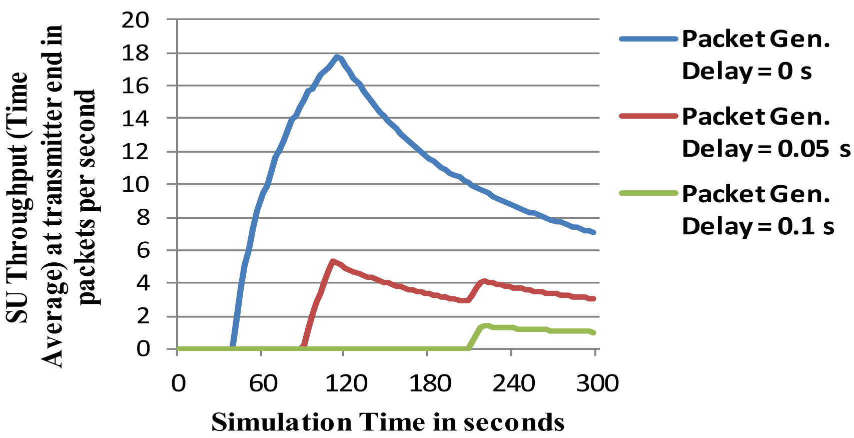
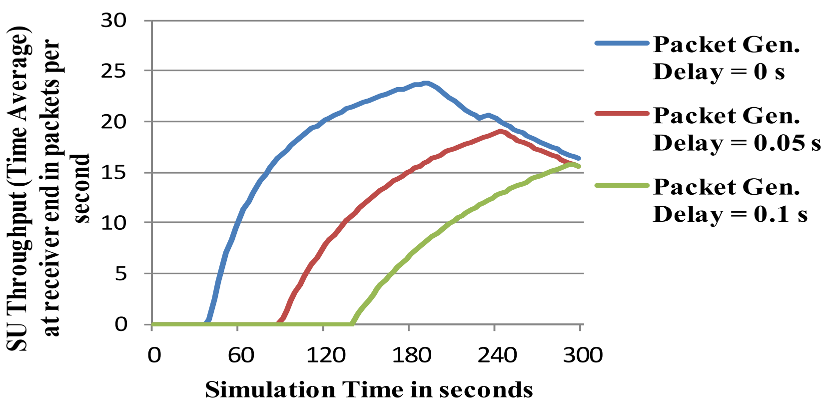


Fig. 3.24 Variation in SU Transmission Time and corresponding SU interference with PU for varying packet arrival delays



(a)



(b)

Fig. 3.25 Variation in SU throughput for different packet generation delays (a) at transmitter end, and (b) receiver end

Similarly, the effect of packet reception delay is witnessed in Fig. 3.26 which reflects considerable decrease in throughput at the receiver end as late arrival packets are often discarded for VoIP sessions. On the whole, every packet induced delay must be minimized to retain the overall QoS of VoIP in CRN.

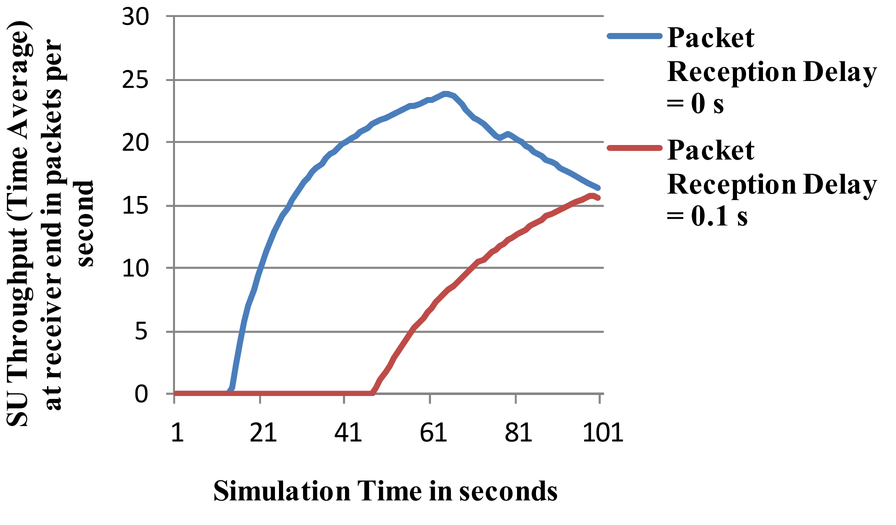


Fig. 3.26 Variation in SU throughput at receiver end for different packet reception delays

3.5 Design of Simulation Model in Visual C++

Simulation model design is further extended in this section following the centralized architecture, where a simple CRN involving primary and secondary users in a single channel scenario is created in Visual C++ [3.40]. Visual C++ has been chosen for this work as it is the foremost language used in simulation today [3.41]. Network domain has already witnessed simulation studies in C++ as reflected in [3.42-3.44]. C++ has also observed profound use with respect to CRN as in [3.10] and [3.45]. Moreover, it is a generic platform, that has been fathered, matured and organized in an interstitial environment and involves dis-embedding and re-embedding of its generic features in specific applications that involves intermittent selective boundary crossings [3.41] and thus facilitates cross-layer modeling strategies. Accordingly, the model overview is presented in the following section that is studied subsequently to analyze the performance of VoIP SUs.

It is worth mentioning that the model design is performed both in OPNET Modeler 16.0.A and Visual C++, as each simulation model offers some unique advantages. This is broadly discussed in the later sections.

3.5.1 Model Overview

Both the PUs and SUs are modeled based on the basic principle of CR cycle as shown in Fig.3.1. Briefly, the PU is the licensed user who has the priority to use the channel. But it does not always occupy the channel, which leads to the channel being underutilized in the time domain. The SU is CR user who is permitted to use the channel only in the absence of any PU. The SU in this model senses the channel during a sensing period and starts its transmission in its transmission period only when the PU is inactive.

The basic modules of the developed model are described as follows.

- *long int* genNumber (long int time)*: This module determines the primary user traffic distribution. It generates the timestamps that mark the onset of primary user traffic arrival. The timestamp generation is dependent on primary traffic distribution that may be, for example, Poisson distribution [3.46], Pareto distribution [3.47] or Markov Modulated Poisson Process (MMPP) [3.48] distribution.

- *int genPacket (int time)*: This module calculates the primary transmission time. Depending on applications and traffic distributions, the output will vary. A VoIP application, for example, will generate spurts of packets at a time followed by periods of silence.

- *void allocSec (int time)*: This module allocates transmission time for the SU. It is assumed in this model that there are always SUs available for transmission as this aims to maximize the channel utilization [3.30] and fulfill the overall objective of CRN.

- *int checkPri (long int j, long int t_time, long int *flag_random, long int *num_random)*: This module is the most significant module and is involved in sensing of the arrival of PU and taking appropriate action. It has several sub-modules that, in turn, sense the presence of PU activity and allocate time slots accordingly.

- *void showMetrics ()*: This module displays the metrics calculated as per the algorithm. It shows the parameters of both CRN and VoIP domain. The most significant parameters include total number of packets lost with respect to primary and secondary users, total sense time elapsed, total PU transmission time elapsed, total SU transmission time, total number of incoming packets (both for PUs and SUs) at the buffer, obtained throughput, etc.

- *void main()*: It serves as the initial and final point of execution of this model execution. Apart from calling the appropriate modules, it maintains the status of every time slot and calculates the metrics accordingly. It also maps the normal and active queue management policies in the model and maintains the queue size accordingly.

This model is based on centralized client-server based architecture in order to facilitate VoIP deployment. Also, a cross-layer architecture is followed in this model where spectrum management decisions work in tandem with the communication protocols [3.30]. The sense and transmit processes, respectively, sense and transmit packets according to the principle stated beforehand. Sensing is performed via single radio architecture [3.32] whose advantages are already discussed in Section 3.3.2.

3.5.2 Study of Simulation Data

Simulation is carried out in the designed model for the analysis of VoIP performance over CRN. The primary difference with the simulation analysis of the previous sections is the role played by the buffers of the centralized networks in these CR systems. Accordingly, it is observed from Fig. 3.27 that increase in sensing interval and decrease in secondary transmission interval increase the number of outstanding packets for SUs in buffers. Therefore, sensing period must have a maximum limit to avoid packet loss due to buffer overflow. However, delay increases thereby degrading the VoIP call quality.

The scenario varies from the PU perspective. For low PU activity, it is noticed that the number of sensing intervals where PU is absent or goes undetected is more. These are wasted time intervals for SUs as they abstain from VoIP transmission. This scenario results in the creation of outstanding SU packets in buffers. The reason is that if the sensing intervals are kept large and PU is detected, SUs can perform spectrum handoff and avoid the "stay on" policy in the current channel. It is observed from Fig. 3.28 that wasted time intervals increase with decrease in sensing interval especially when PU activity is low.

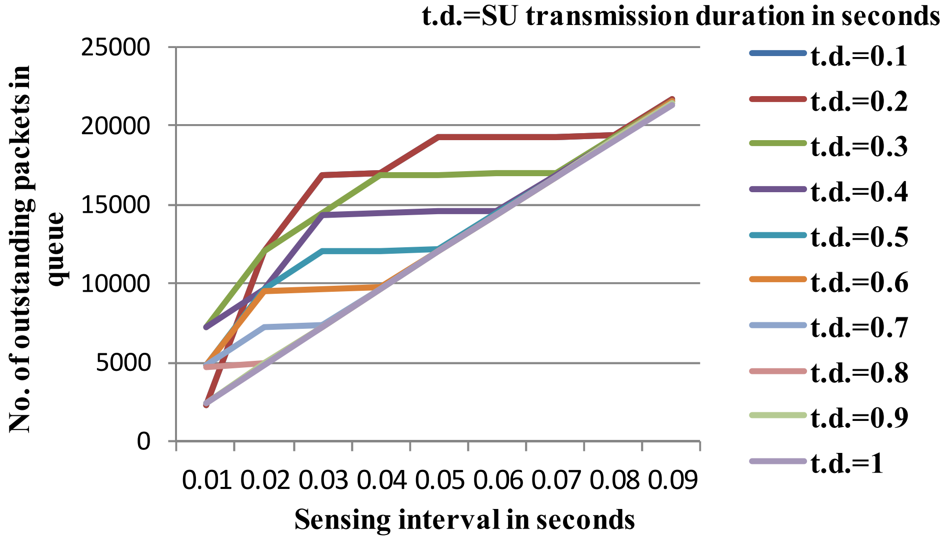


Fig. 3.27 Variation in number of SU packets waiting to be transmitted with different sensing and transmission intervals

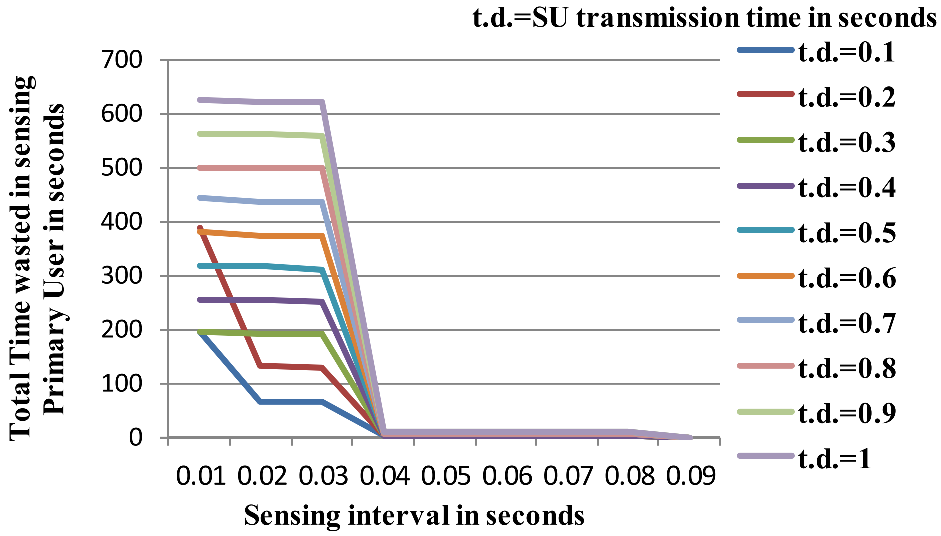


Fig. 3.28 Variation in sensing time wasted for PU detection for different SU sensing and transmission intervals

It is further noticed from Fig. 3.29 that the effective transmission time of SU, free from any PU interference, is high for low sensing intervals. However, this duration is not directly proportional to SU transmission time interval. This is due to the fact that increase in effective (actual) transmission time for SU is lesser compared to the configured transmission interval due to the effect of increased PU interference.

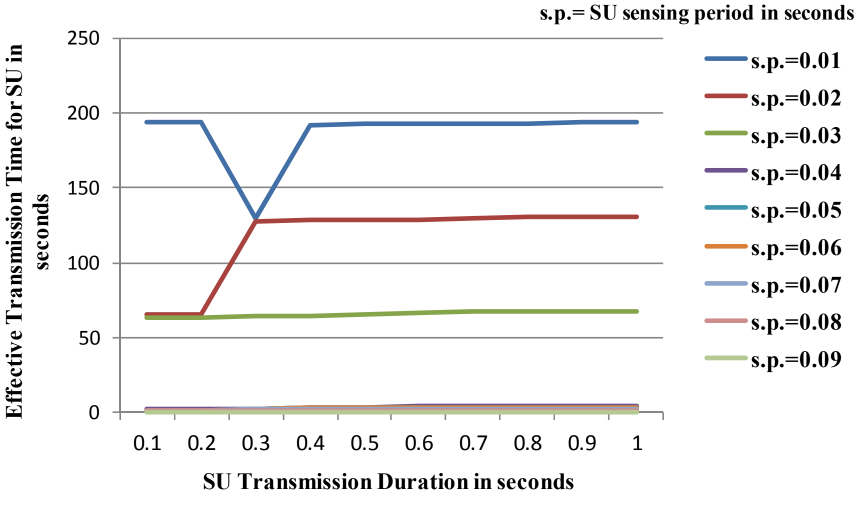


Fig. 3.29 Variation in effective transmission time for SUs with different sensing and transmission intervals of SUs

Thus, this model provides us with a different perspective of the CRN where a centralized architecture is utilized to analyze the effect of buffers on the performance of VoIP based SUs. Also, this model will also serve as the platform for implementing the proposed codec adaptation algorithm to be explained later in Section 3.7.

3.6 Mathematical Formulation

Once the simulation model is designed and analyzed in Sections 3.3 and 3.5, the outcome must be validated through the formulation of a suitable mathematical model. However, following the principles of prototype modeling, such a mathematical model must be developed initially with basic approximations and gradually that model must be refined to present a more realistic approach. With respect to VoIP over CRN, a detailed mathematical analysis must cover all critical factors ranging from timing parameters (including sensing time, transmission time), PU and SU activities, attributes pertaining to CRN network, VoIP codec parameters, queuing attributes, etc. However, such an analysis is implementation specific and is also dependent on the prevailing network conditions and user activity patterns. As this chapter primarily aims to provide the initial platform for model development and subsequent research in the domain of VoIP over CRN, a basic mathematical model based on the total time of VoIP transmission is presented in this section that forms the basis of the proposed simulation models. The preliminary analysis is based on the consideration that sensing and transmission time durations are uniform throughout the VoIP communication duration. Moreover, spectral handoff is not considered in this work as it depends on various implementation strategies and spectral handoff policies (proactive and reactive) [3.35] and are addressed in the later chapters.

Let T be the total time of VoIP transmission that includes the sensing time t_s and SU transmission time t_d. Let P_{si} and P_{di} denote the probability of PU arrival in ith sensing period and ith transmission period respectively. The expression for T is derived as follows.

Total elapsed time $T = \sum_{over\ all\ cycles}$ [for each cycle if PU does not arrive in the previous transmission cycle: *sensing time* + *transmission time* (if the channel is detected idle in the sensing time)]

$$\Rightarrow T = \left[t_{s1} + (1 - P_{s1})t_{d1}\right] + (1 - P_{d1})\left[t_{s2} + (1 - P_{s2})t_{d2}\right] \\ + (1 - P_{d2})\left[t_{s3} + (1 - P_{s3})t_{d3}\right] + \ldots + (1 - P_{dn-1})\left[t_{sn} + (1 - P_{sn})t_{dn}\right] \tag{3.2}$$

Simplifying (3.2), the expression for T is given by,

$$T = \sum_{i=1}^{n} t_{si} - \sum_{i=1}^{n-1} P_{di} t_{si+1} + \sum_{i=1}^{n} (1 - P_{si}) t_{di} \\ + \sum_{i=1}^{n-1} P_{di} \left(1 - P_{s(i+1)}\right) t_{d(i+1)} \tag{3.3}$$

Considering fixed t_{si} and t_{di} time durations $\forall\, i's$ and simplifying (3.3), the expression for T is obtained as follows.

Sensing Block

$$T = t_s\left[n-(n-1)\sum_{i=1}^{n-1}P_{di}\right]$$

Transmission Block

$$+\,t_d\left[n\sum_{i=1}^{n}(1-P_{si})+(n-1)\sum_{i=1}^{n-1}P_{di}\left(1-P_{s(i+1)}\right)\right] \tag{3.4}$$

It is apparent that PU arrival probabilities, namely P_{si} and P_{di} are interdependent as described below. Let λ be the PU arrival probability which is considered to be uniform for every time slot in this chapter. (It is to be noted that when the model is refined for advanced research and learning, λ will be governed by the probability distribution function of the corresponding PU traffic distribution.) Accordingly,

$$\begin{aligned}
P_{d1} &= \lambda(1-P_{s1}) \\
P_{s2} &= \lambda(1-P_{d1}) = \lambda\{1-\lambda((1-P_{s1}))\} \\
P_{d2} &= \lambda(1-P_{s2}) = \lambda[1-\lambda\{1-\lambda(1-P_{s1})\}] \\
&\vdots \\
P_{sn} &= \lambda(1-P_{d(n-1)}) = \lambda[1-\lambda\{1-\lambda\{\ldots.\,\lambda(1-P_{s1})\}\}] \\
P_{dn} &= \lambda(1-P_{sn}) = \lambda[1-\lambda\{1-\lambda\{\ldots.\,\lambda(1-P_{s1})\}\}]
\end{aligned} \tag{3.5}$$

Therefore, P_{si} and P_{di} for the ith time slot can be derived and expressed as follows.

$$P_{si} = \lambda(1+\lambda)\frac{\left[1-\lambda^{2i-2}\right]}{1-\lambda^{2}} + \lambda^{2(i-1)}P_{s1} \tag{3.6}$$

$$P_{di} = \lambda\frac{\left[1+\lambda^{2i-1}\right]}{1-\lambda^{2}} - \lambda^{2i-1}P_{s1} \tag{3.7}$$

Thus, the general expression for T is derived from (3.4) and expressed in (3.8). It is clearly observed from (3.8) that VoIP transmission time in the basic scenario depends on PU traffic characteristics that govern the PU arrival probability λ. Furthermore, as inferred from (3.8), selection of optimal sensing and transmission durations also plays an important role towards deciding the successful VoIP communication duration and is, therefore, studied in the next chapter.

$$T = t_s\left[n-(n-1)\sum_{i=1}^{n-1}\left\{\lambda\frac{\left[1+\lambda^{2i-1}\right]}{1-\lambda^2}-\lambda^{2i-1}P_{s1}\right\}\right]$$
$$+t_d\left[n\sum_{i=1}^{n}\left\{1-\left\{\lambda(1+\lambda)\frac{\left[1-\lambda^{2i-2}\right]}{1-\lambda^2}+\lambda^{2(i-1)}P_{s1}\right\}\right\}\right.$$
$$\left.+(n-1)\sum_{i=1}^{n-1}\left[\left\{\lambda\frac{\left[1+\lambda^{2i-1}\right]}{1-\lambda^2}-\lambda^{2i-1}P_{s1}\right\}\left\{1-\left\{\lambda(1+\lambda)\frac{\left[1-\lambda^{2i}\right]}{1-\lambda^2}+\lambda^{2i}P_{s1}\right\}\right\}\right]\right] \tag{3.8}$$

3.7 Proactive Codec Configuration for VoIP Communication

While discussions are made on the effect of different network and system level characteristics (such as sensing and transmission durations, activity patterns, buffer overflows, packet processing delays, etc.) to determine the success of VoIP transmission over CRN, the application–specific aspects of VoIP are not yet studied. This section therefore performs an application-oriented study where the VoIP parameters such as codecs are specifically addressed and configured them using a novel algorithm.

Codec adaptation algorithms for implementation of VoIP service over specialized networks like CRN is an area that requires investigation to ensure better spectral utilization and support for more calls. It is clearly evident from the literature survey in Section 3.1 that tuning of codec parameters in relation to varying network dynamics under different PU traffic activities is yet to be studied extensively. Accordingly, this section designs a novel codec adaptation algorithm using active queue management for VoIP based SUs and implements it in the designed Visual C++ based model (as already explained in Section 3.5).

At first, an overview of the codec properties and active queues is provided followed by subsequent design and implementation of the proposed algorithm.

3.7.1 Background Study

(i) Overview of Codecs

A codec [3.49], which stands for coder-decoder, converts an audio signal into compressed digital form for transmission and then back into an uncompressed audio signal for replay. Codecs accomplish the conversion by sampling the audio signal several thousand times per second. It converts each tiny sample into digitized data and compresses it for transmission. Codecs use advanced algorithms to help sample, sort, compress and packetize audio data. The CS-ACELP algorithm [3.50] (Conjugate-Structure Algebraic-Code-Excited Linear Prediction) is one of the most prevalent algorithms in VoIP.

Every codec is defined by some parameters that guide their performance, the most significant among them being *codec bit rate (ε)*. Based on codec, this is the number of bits per second transmitted and is given by (3.9) [3.49].

$$\varepsilon = \frac{\upsilon}{\delta} \tag{3.9}$$

where δ is the codec sample interval and υ is the codec sample size, that is, the number of bytes/bits captured by Digital Signal Processor (DSP) at each codec sample interval.

Mean Opinion Score (MOS) is another important parameter and a system of grading voice quality of telephone connections. It has thoroughly been explained in Section 2.6 in Chapter 2.

The *voice payload size (s)* is also another significant parameter, which represents the number of bytes (or bits) that are filled into a packet. It must be a multiple of the codec sample size. *Packets per second (pps)* [3.49] is the most important parameter. It represents the number of packets transmitted every second in order to deliver the codec bit rate and is given by (3.10) [3.49].

$$pps = \frac{\varepsilon}{s} \tag{3.10}$$

One of the important factors to consider while building packet voice networks is proper capacity planning. Within capacity planning, *Bandwidth (B)* calculation [3.49] is an important aspect and is given by,

$$B = total \quad packet \quad size \times pps \tag{3.11}$$

where total packet size = (L2 header: MP or FRF.12 or Ethernet) + (IP/UDP/RTP header) + (voice payload size).

(ii) ***Overview of Active Queues***

Active queues drop packets before the queue is full to ensure fairness among multiple users and avoid loss in throughout. The Random Early Detection (RED) algorithm [3.51] is one of the active queue management policies. It has two parts namely,

1. *Estimation of average queue size*

The average queue size is given by (3.12) as follows.

$$Qavg = (1 - W) \times Qavg + W \times Qsample \,, 0 < W < 1 \tag{3.12}$$

where, Q_{avg}= average queue size;

Q_{sample}= instantaneous queue size;

W= weight;

2. *Decision to drop packet*

The decision to drop the packet must depend on the average and not the instantaneous queue size. Two thresholds are defined as Min_{th} and Max_{th}. The decision to drop a packet is guided by the following conditions.

- If $Min_{th} < Q_{avg} < Max_{th}$, packet is dropped with a probability p_{queue}.
- If $Q_{avg} < Min_{th}$, no packet is dropped.
- If $Q_{avg} > Max_{th}$, every packet is dropped.

The probability p_{queue} is given by the following expression.

$$p_{queue} = \max \; p_{queue} \left\{ (Q_{avg} - Min_{th}) / (Max_{th} - Min_{th}) \right\} \tag{3.13}$$

So the selection of appropriate threshold limits is crucial for successful implementation of active queue management system to ensure significant performance improvement in the networks.

Based on the codec properties and active queues, the objective in this work is to decrease the *pps* parameter without significant delay and conserve more bandwidth in the congested network scenarios. As number of users increase in CRN, there is bottleneck in the centralized APs. Therefore, increasing *pps* will further add to packet loss due to buffer overflow and must be reduced. Additionally, active queues must also be implemented in order to follow a proactive strategy for QoS enhancement in VoIP calls. Considering all these factors, a novel codec adaptation algorithm is proposed in the following section.

3.7.2 Design of the Proposed Algorithm

The idea is to implement a proactive strategy to adaptively maintain codec bit rate by implementing active queue management. The buffer capacity is analyzed and based on certain estimations about possible future network conditions, the decisions regarding the switching of codec bit rate are taken.

Active queue management based proactive QoS configuration requires critical decisions with respect to buffer capacity. Therefore, *Queue Occupancy (Q.O.)* factor is introduced in this section to assist in taking such decisions. Also, few issues have been assumed to be already present at the onset of algorithm implementation. First and foremost, switching of lower bit rate codec to higher bit rate codec and vice versa is done based on the Sender Reports (SR) and Receiver Reports (RR) [3.52] that are created from network loss and delay. Therefore, it is assumed that the softphone can switch codecs based on SR and RR. Thereafter, the QoS metrics for ascertaining the performance of the network namely delay, jitter, packet loss, MOS and R-Factor are categorized into *good*, *tolerable* and *poor* limits. Beyond the *good* limit is considered as the threshold limit which is denoted by *thresh*.

The proposed algorithm is comprised of three parts namely, 1) implementation of active queue management, 2) adaptive variation of bit rates, and 3) codec configuration in CRN. All these parts are implemented simultaneously.

(i) Implementation of Active Queue Management

a) Calculation of Queue Occupancy Factor

Queue Occupancy (Q.O.) is defined as the factor that guides the configuration of the RED [3.51] buffer parameters in APs. *Q.O.* indicates the extent to which the buffer is occupied for a certain time interval and is based on a scale of 1 to 10. A value of more than 5 indicates that the queue is likely to get filled up soon and a value of less than 2 indicates that the buffer will not be full in the near future. It is an approximation and may vary from actual result. Let the total queue size be N. *Q.O.* is calculated as follows.

Module 1.a: Calculation of Queue Occupancy Factor	
Step 1:	Monitor the number of packets that are currently in the queue at instant t_1. Let it be n_1.
Step 2:	Monitor the number of packets that are currently in the queue at instant t_2. Let it be n_2. The time interval $t=t_2-t_1$ is set beforehand.
Step 3:	The rate of occupancy r is calculated as $r = (n_2-n_1)/t$. A negative value suggests decrease in packets in the queue.
Step 4:	The expected arrival of packets N_1 is calculated as, $N_1 = (n_2+r*t)*p$ where p is the packet size.
Step 5:	*Queue Occupancy* factor is calculated as $Q.O.=N_1/N*10$.

b) Implementation of RED Buffer as an Active Queue

Module 1.b: Implementation of RED Buffer	
Step 1:	Monitor *Q.O.* factor as described in *Module 1.a*.
Step 2:	If *Q.O.* < 2 then go to *Module 2*.
Step 3:	If 2< *Q.O.* <5 then implement RED queue with Min_{th}=50 and Max_{th}=100, where Min_{th} *and* Max_{th} denote minimum and maximum threshold limits respectively.
Step 4:	If *Q.O.*>5 then implement RED queue with Min_{th}=70 and Max_{th}=100.

Step 5:	Go to *Module 2*.
Step 6:	If call does not terminate, wait for a time interval and go to *Step 1*.

(ii) Adaptive variation of bit rates

Module 2: Adaptive Variation of bit rates	
Step 1:	Monitor packet loss ($loss_{pac}$) in the network.
Step 2:	If $loss_{pac}$ >*thresh* switch to lower bit rate. Go to Step 4.
Step 3:	Switch to higher bit rate.
Step 4:	Go to step 6 of *Module 1.b*.

A pictorial representation of the two modules is depicted in Figure 3.30.

(iii) Codec Configuration in CRN

Module 3: Codec configuration in CRN	
Step 1:	Check the status of SU arrival. If it arrives during the sensing period, set *status*=1 else set *status*=2.
Step 2:	Check the status of PU arrival. If it arrives during the sensing period, set *pr_status*=1 else set *pr_status* =2.
Step 3:	If *status* =1, start VoIP transmission with low codec bit rate, else start VoIP transmission with high codec bit rate.
Step 4:	If *status* =2 and *pr_status*=2 and current codec bit rate is high, change to low codec bit rate.
Step 5:	If *status*=2 and *pr_status*=1, calculate *Q.O.* as described in *Module 1* and change the codec bit rate accordingly by following the general procedure in *Module 2*.

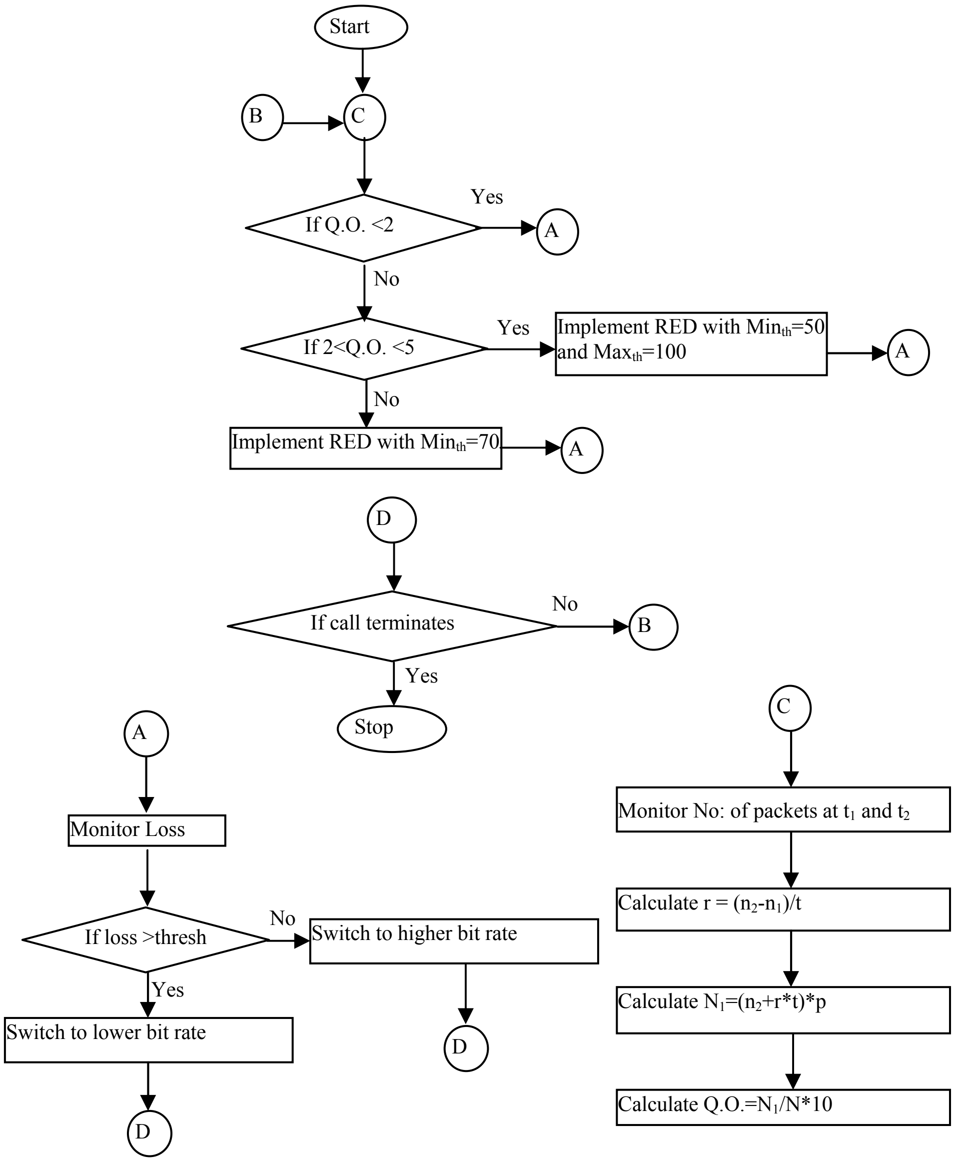


Fig. 3.30 Flowchart depicting the first and second modules of our proposed codec adaptation algorithm

3.7.3 Implementation of the Proposed Algorithm

The complete algorithm is implemented in the developed model (that has been developed in Section 3.5). Initially in the basic CRN scenario, it is clearly observed from Fig. 3.31 and Fig. 3.32 that the throughput of the SU is less for low bit rate codecs in comparison to the high bit rate codecs in CRN.

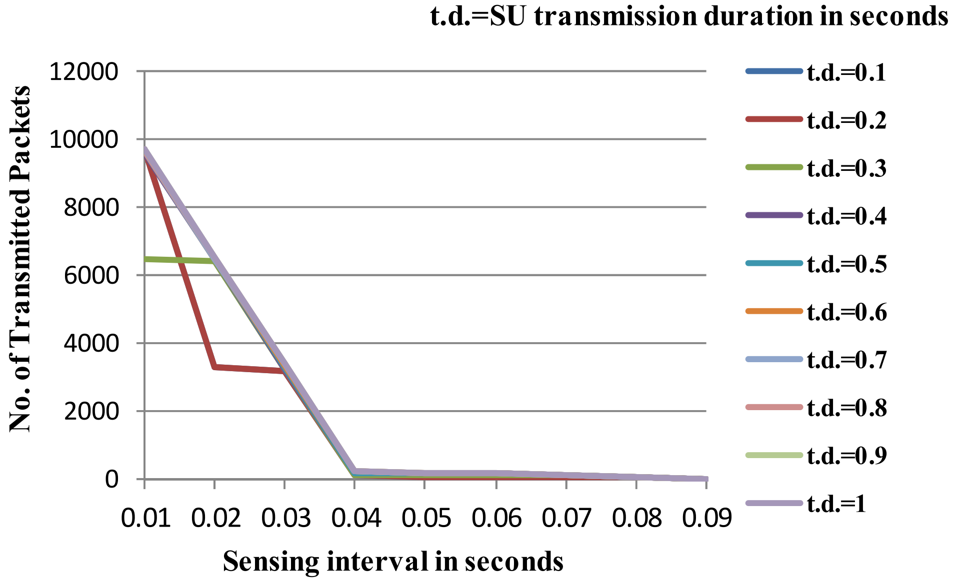


Fig. 3.31 Variation in SU throughput for different sensing and transmission intervals with respect to high bit rate codecs

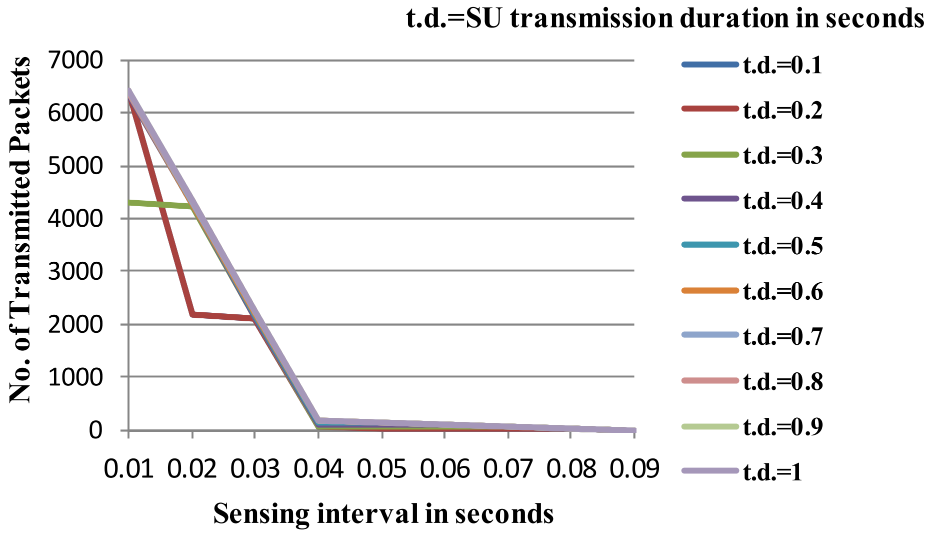


Fig. 3.32 Variation in SU throughput for different sensing and transmission intervals with respect to low bit rate codecs

However, after implementation of the proposed algorithm, Fig. 3.33 shows the variation in throughput of the SU which records the maximum value compared to the previous cases. Thus, the algorithm succeeds in providing the maximum throughput without interference from the PU, as compared to Fig. 3.31 and Fig. 3.32.

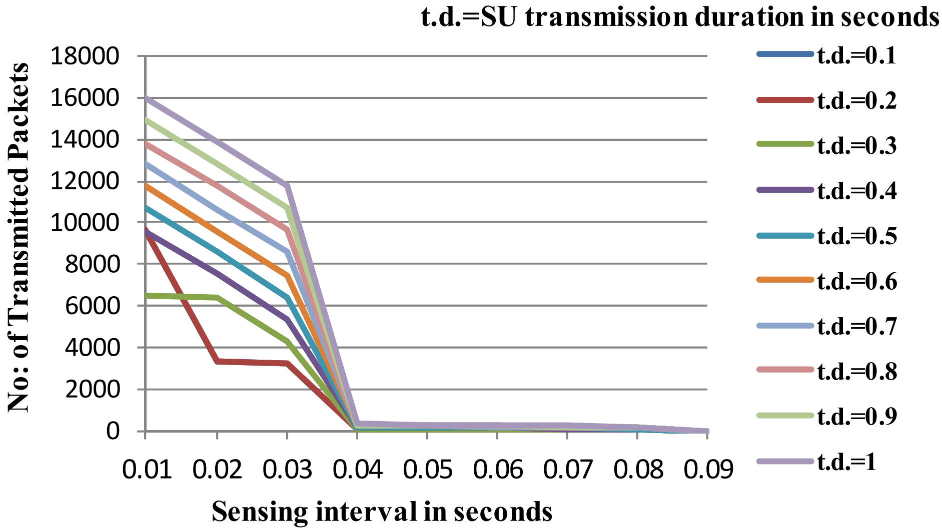
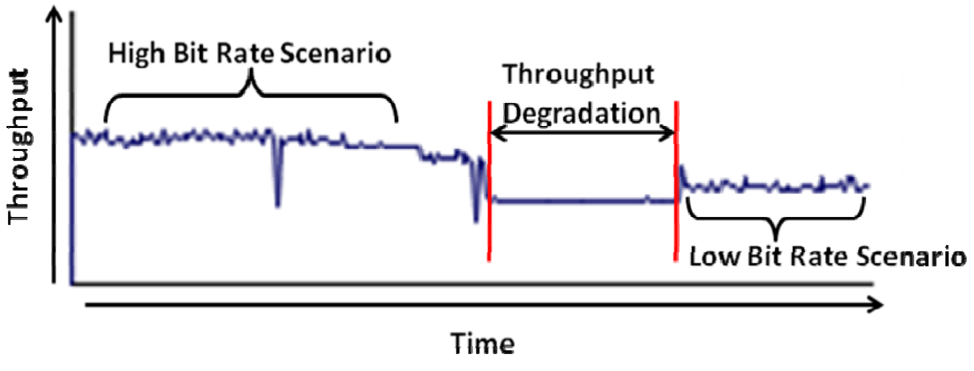


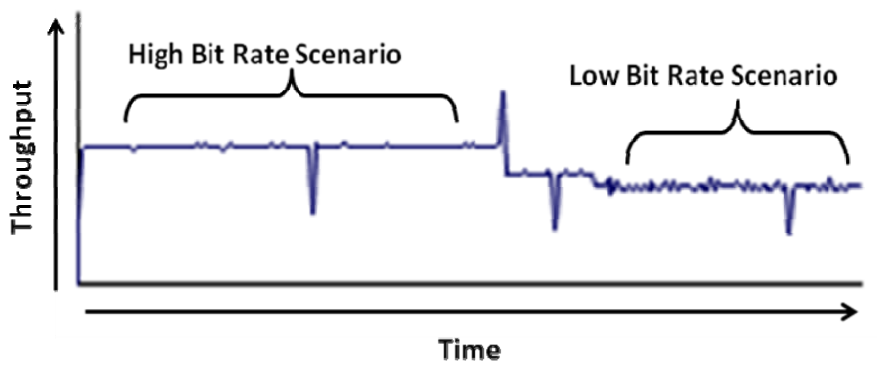
Fig. 3.33 Variation in SU throughput for different sensing and transmission intervals after implementation of the proposed algorithm

Moreover, with the implementation of fixed buffer, throughput degrades as packet loss increases due to buffer overflow. This is because switching to lower bit rate to decrease *pps* is done after the queue is full. As a result, packet loss decreases after a certain tine interval only when the queue starts getting empty. Fig.3.34 (a) records this degradation in throughput. On the other hand, RED implementation in the algorithmic scenario ensures that the switching to lower bit rate is done before the queue is full. This proactive approach of keeping the buffer partially empty avoids packet loss due to buffer overflow (and hence throughput degradation) as observed in Fig. 3.34 (b).

However, since SU transmits VoIP traffic in alternate sensing and transmission intervals, even this enhanced throughput quickly degrades with a minimal increase in the sensing interval. This establishes the fact that apart from configuration of VoIP parameters, appropriate tuning of the CRN parameters also plays a crucial role in ensuring the success of VoIP communication over CRN.



(a)



(b)

Fig. 3.34 Variation in throughput with time for (a) low bit rate fixed buffer scenario, and (b) our proposed algorithm

3.7.4 Discussions

Let $Throu_{hbr}$, $Throu_{lbr}$ and $Throu_{pro}$ be the expected throughput with respect to SU voice traffic after application of high codec bit rate, low codec bit rate and adaptive codec bit rate (corresponding to the proposed algorithm) respectively. Let the *pps* corresponding to high, low and adaptive codec bit rate be α_{s_hbr}, α_{s_lbr} and α_{s_abr} respectively. Also, let P_{pt} and P_{ps} be the probabilities that the PU traffic arrives during the SU transmission interval and during the sensing intervals respectively.

When PU arrives in the licensed channel currently occupied by the SU, the PU can arrive in either the sensing or the SU transmission time interval. Hence for a single CR cycle,

$$P_{ps} + P_{pt} = 1 \tag{3.14}$$

The throughput for the SU traffic $Throu_{sec}$ is defined as,

$$Throu_{sec} = Total\ SU\ packets\ generated\ - Total\ SU\ packets\ lost\ due\ to\ interference\ with\ the\ PU \quad (3.15)$$

Deriving the expressions for $Throu_{hbr}$, $Throu_{lbr}$ and $Throu_{pro}$ based on (3.15),

$$Throu_{hbr} = \alpha_{s\_hbr} \times T_d \times (1 - P_{pt}) \quad (3.16)$$

$$Throu_{lbr} = \alpha_{s\_lbr} \times T_d \times (1 - P_{pt}) \quad (3.17)$$

$$\begin{aligned} Throu_{pro} &= (T_d \times \alpha_{s\_hbr}) - (T_d \times \alpha_{s\_lbr} \times P_{pt}) \\ &= T_d \times \{(\alpha_{s\_hbr}) - (\alpha_{s\_lbr} \times P_{pt})\} \end{aligned} \quad (3.18)$$

(where T_d is the transmission interval).

Replacing the value of P_{pt} from (3.14) in (3.18),

$$Throu_{pro} = T_d \times [(\alpha_{s\_hbr}) - \{\alpha_{s\_lbr} \times (1 - P_{ps})\}] \quad (3.19)$$

Subtracting (3.17) from (3.19),

$$Throu_{hbr} - Throu_{lbr} = T_d \times (1 - P_{pt}) \times (\alpha_{s\_hbr} - \alpha_{s\_lbr}) \quad (3.20)$$

As $\alpha_{s\_hbr} > \alpha_{s\_lbr}$, it is clear from (3.20) that

$$Throu_{hbr} > Throu_{lbr} \quad (3.21)$$

However, considering the scenario where the PU has increased probability of arriving during the SU transmission period, P_{pt} increases. Hence from (3.16) and (3.17), it is observed that both $Throu_{hbr}$ and $Throu_{lbr}$ decreases. Thus with increased PU activity, the VoIP throughput for SU decreases for both high and low bit rate codecs.

Further, subtracting (3.16) from (3.18),

$$\begin{aligned} Throu_{pro} - Throu_{hbr} = T_d \times [(\alpha_{s\_hbr}) - \{\alpha_{s\_lbr} \times (1 - P_{ps})\} \\ - \{\alpha_{s\_hbr} \times (1 - P_{pt})\}] \end{aligned} \quad (3.22)$$

Replacing the value of P_{pt} from (3.14) in (3.22),

$$Throu_{pro} - Throu_{hbr} = T_d \times [(\alpha_{s\_hbr}) - \{\alpha_{s\_lbr} \times (1 - P_{ps})\}$$

$$- \{\alpha_{s\_hbr} \times P_{ps}\}]$$

$$= T_d \times \{(\alpha_{s\_hbr} - \alpha_{s\_lbr}) \times (1 - P_{ps})\} \tag{3.23}$$

From (3.23), it is observed that

- $T_d > 0$ as time cannot have negative value.
- $1 - P_{ps} \geq 0$ as the theoretical maximum value of any probability is 1.
- $\alpha_{s\_hbr} > \alpha_{s\_lbr}$ due to increased codec bit rate.

Therefore,

$$Throu_{pro} - Throu_{hbr} \geq 0$$

$$\Rightarrow Throu_{pro} \geq Throu_{hbr} \tag{3.24}$$

It is, hence, verified from (3.21) and (3.24) that $Throu_{pro}$, that is, the throughput achieved after implementation of the proposed algorithm, is the maximum throughput achieved and this proves the efficiency of the proposed algorithm and validates the observations in Fig 3.34. As the throughput obtained is free from any interference with the PU, it can also be concluded that the number of outstanding packets in the AP queue (and hence the packet loss incurred due to buffer overflow) is minimum. This also implies minimal queuing and medium access delays for the packets. Thus the algorithm enhances the VoIP call quality with respect to the SUs.

3.8 General Discussions on the Designed Simulation Models

Finally, in this section, the advantages of the designed simulation models are highlighted. Also, the simulation results from the different designed models in Section 3.3 and 3.5 are validated through comparative performance evaluation.

3.8.1 Benefits of the Design Approach in OPNET Modeler 16.0.A

The generic mechanism for the development and subsequent maintenance of simulation model requires software engineering concepts along with application expertise. Hence, the Prototype model [3.53] has been implemented in this design workflow. This approach provides a crude system model that serves as the focal point for initial analysis.

Moreover, multi-paradigm modeling [3.54] is followed while designing models in OPNET Modeler 16.0.A. The system is considered to be a collection of objects with processes initiating state changes. The advantage is that parallel processing of the object behavior is ensured that is mandatory in simulations involving long runs [3.1]. However, OPNET Modeler 16.0.A supports conservative approach of system modeling [3.55] with potential for deadlock and hence, care is taken to prevent it.

The developed simulation models can be used for analysis of the existing algorithms in the field of VoIP performance in CRN, experimentation and application of new algorithms with appropriate modifications to the developed model and as training tools to acquaint researchers with ideas of VoIP implementation in CRN. The simulation output can be analyzed quantitatively and qualitatively along with post simulation animation [3.56] that visualizes input, internal and output behavior of designed models.

3.8.2 Advantages of the Designed Model in Visual C++

Visual C++ based model design offers some unique advantages as well. Firstly, the model design is modular in nature. Modularity increases simulation speed and allows debugging of errors. Moreover, such a design is encouraged as a valid and simpler model is efficient [3.57]. Moreover, in the absence of any standard for VoIP implementation over CRN, the principles of conceptual modeling [3.58] are followed in this model. Lack of knowledge hinders design of simulation model. Hence, the model is run iteratively to gather data and use it for modifications.

Advantages of adopting an iterative model are stated as follows.

- It avoids designers from sticking to a certain model, which is one of the serious perils of simulation design [3.59].
- It allows the scope of introducing agent based modeling [3.54] where the system gradually evolves from interaction of agents.
- Increase in simulation runs with incremental model design detects possible errors and discrepancies in the model [3.57].

In addition, as the lack of standard data for VoIP over CRN proves to be a major obstacle towards proper model design, statistical analysis of limited simulation data is not encouraged as such data is easy to misinterpret resulting in gross error [3.60] and inefficient modeling. Rather, analysis of simulation data in this model is done graphically as it is simpler and easier to interpret and provides useful information about the overall trend in the variation of parameters.

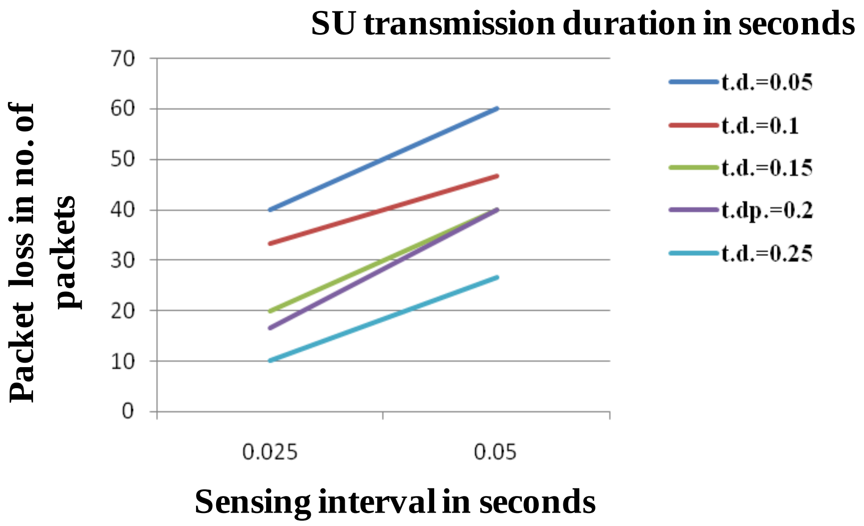
Finally, this model incorporates the effect of buffer overflow in a centralized CRN architecture and thus provides the researchers with a prototype to implement further optimizations in this domain.

Overall, it can be inferred that the simulation models in OPNET Modeler 16.0.A are built using customizable functions and parameters. While it gives enough flexibility to the users in terms of design and implementation, the process is highly complex and requires repeated compilations, which consume time. Visual C++ is simpler and allows easier and quick customizations but at the cost of less efficient coding. Hence, the simpler concepts of centralized architecture are designed in Visual C++ by applying optimized coding techniques. Distributed architecture requires complex coordination between nodes and hence, OPNET Modeler 16.0.A is selected to implement such principles and develop the models accordingly.

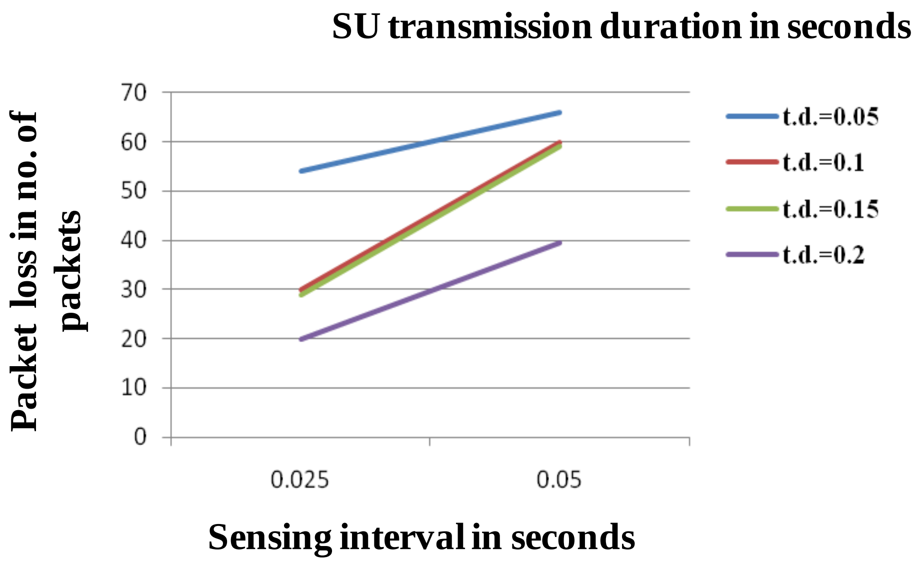
3.8.3 Validation

Validation of the developed models is performed by comparison of simulation data obtained from each model. It must be noted that as these platforms behave completely differently, it is very difficult to compare their absolute values. On the other hand, observing the trends in these simulation results indirectly verify their research outcome as already evident in this chapter.

Nonetheless, identical simulation runs are conducted in these models to further establish the credibility of their outcome. Fig. 3.35 (a) and Fig. 3.35 (b) show the output obtained from the models in OPNET Modeler 16.0.A and Visual C++ respectively for similar input data.



(a)



(b)

Fig. 3.35 Variation in packet loss for SU with different sensing and transmission intervals in models developed in (a) OPNET Modeler 16.0.A, and (b) Visual C++

It is observed from both the scenarios that increase in sensing period increases SU packet loss due to unwanted disruptions in VoIP transmissions. Also, packet loss is inversely proportional to the time of VoIP transmission and hence decreases with shorter transmission durations for SUs. Similar trends in simulation output validate that the designed models provide credible results and can be used with confidence for future research studies.

3.9 Summary

This chapter has dealt with the design of VoIP model in CRN under various simulation platforms followed by extensive analysis of the simulation output. The problem domain is defined along with the basic principle of VoIP implementation in CRN. Initially, VoIP models in CRN domain for single and multi-channel scenarios are designed in OPNET Modeler 16.0.A. following distributed architecture. In the next phase, the principle of centralized architecture is applied to develop the model for VoIP communication over CRN in Visual C++. Detailed analysis of simulation output in each scenario indicates strong correlation between VoIP QoS metrics and CRN parameters. Comparison of the corresponding simulation output generated from these models validates the proper design of the models. Performance study of the models reflects efficient simulation design with sufficient credibility. Furthermore, the basic mathematical policy behind the design has been established that has opened up various issues in both VoIP and CRN domains to be analyzed for future research. Finally, the critical attributes of the developed models are studied in detail with the focus on overall call quality improvement for VoIP transmissions over CRN.

Thereafter, this chapter has addressed the performance enhancement for VoIP applications with respect to the codecs and active queues. The proactive configuration of codec parameters is performed in conjunction with active queue management for improved VoIP performance in CRN. An algorithm is proposed based on proactive strategy that suggests adaptive variation of codec bit rate along with RED implementation in the access points. Simulation results confirm VoIP throughput maximization along with enhanced QoS after implementation of the developed algorithm in CRN.

Overall, these models can be used by learners and researchers to gather basic knowledge of VoIP transmission in CRN. Also, there is enormous scope of developing these models further to apply suitable optimizations in VoIP and CR domain.

The outcome of this study has been published in the ***Journal of Networks'12 (SCI Indexed)*** and ***International Journal of Computer Applications'13 (SCI Indexed)***. Also, this work received the ***Best Paper Award in ACM International Conference CUBE'12*** held in Pune, India and ***has subsequently been published in the Conference Proceedings***.

This Chapter

Highlights the strong correlation between QoS of VoIP calls and various CR parameters and, thereafter, designs proactive codec adaptation algorithms.

Next Chapter

Optimal configuration of CR cycle timing parameters followed by suitable detection of available transmission time before initiating VoIP sessions.

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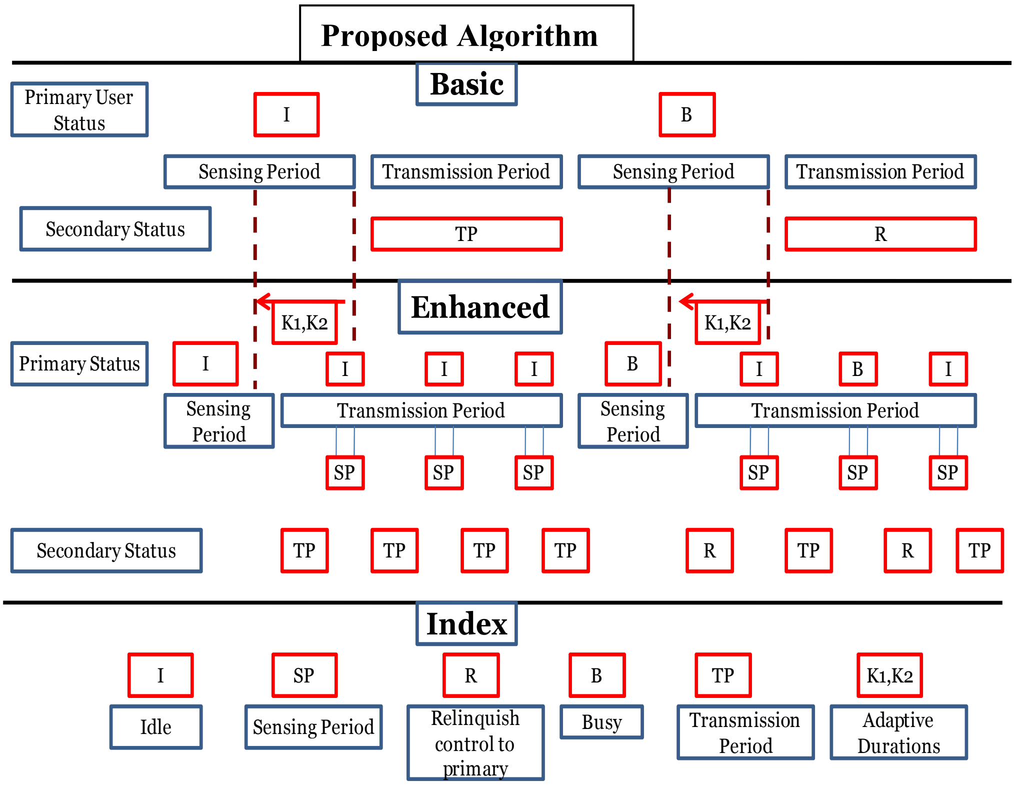
Chapter 4.

ANALYSIS AND ENHANCEMENT OF CR CYCLE TIMING PARAMETERS

Chapter Highlights



CHAPTER 4: Analysis and Enhancement of CR Cycle Timing Parameters

"I started in time-sharing and networking with packet switching, which was the precursor to what became the Internet. Time-shared use on packet-switch networks, when you think about it, is the cloud."

- Audrey McLean, Stanford University

Outline of the Chapter

It is clear from the previous chapter that the basic CR cycle is incapable of successfully hosting the VoIP applications. This is evident from the recorded QoS metrics including end-to-end delay, jitter, packet loss and throughput, all of which witness perceivable degradation when implemented in the simulation models. Therefore, using the simulation models in Chapter 1, this chapter extends the proposed research where it aims to address the QoS degradation problem through the formulation of a QoS aware algorithm that optimally configures the CR cycle parameters comprising of sensing and transmission intervals. Although earlier works in literature have studied this topic and proposed several solutions related to sensing-throughput tradeoff [4.1-4.3] and outage probability calculations [4.4, 4.5], QoS oriented studies with respect to real-time applications such as VoIP are not yet considered during the configuration of sensing and transmission intervals. In this regard, ***the novelty***

of this chapter lies in the efficient design of several QoS-aware strategies that optimally configure the CR cycle parameters and also detect the best possible transmission time before initiating the VoIP session in a particular channel. Using the developed simulation model of Chapter 1 as a platform for evaluation, the proposed strategies are properly implemented and thoroughly analyzed to ensure their credibility in dealing with real-time traffic.

Initially, the CR cycle is modified using the formulation of a novel algorithm that is designed in two parts. The first part introduces momentary sensing slots in the transmission intervals of the SUs. The second part subsequently varies the sensing and transmission slots based on feedback parameters. This is followed by an in-depth study of the design policies in the simulation models under different conditions of PU activity and SU characteristics. Extensive simulation studies reflect overall QoS enhancement for VoIP traffic after application of the modified CR cycle. Mathematical analysis of the algorithm further provides expressions for critical system parameters and also derives necessary conditions for the optimal selection of related parameters (including the sensing time, transmission time, feedback parameters, PU detection probabilities, etc).

Thereafter, the conditions of imperfect spectrum sensing are incorporated in the system model and another algorithm is proposed to select the most efficient transmission duration for initiating VoIP communication, by taking into account the relevant parameters in CRN. Mathematical analysis followed by simulation results bears a clear testimony to the fact that the VoIP call quality is enhanced after implementing this algorithm. Finally, a new QoS parameter namely, *cog_cap* is designed that denotes the cognitive capacity of VoIP calls. Analysis of *cog_cap* for the modified CR cycle indicates high cognitive capability of VoIP calls thereby proving the efficiency of the proposed technique in this chapter.

4.1 Introduction

A basic CR cycle comprises of a sensing interval followed by a transmission interval. As already described in Chapter 2, every SU senses the presence of a PU in the sensing interval using detection algorithms such as energy detection schemes, cyclostationary feature detection [4.6], etc. If the PU is found to be absent, the SU resumes transmission for the entire duration of the transmission period. Obviously, increasing the sensing duration will increase the probability of PU detection at the cost of SU throughput and vice-versa. This sensing-throughput trade-off assumes special significance with respect to VoIP traffic. For example, when the SU is in sensing mode, it abstains from any further transmission resulting in packet loss for the VoIP calls. If buffer is utilized by the application to reduce this loss, it can subsequently lead to increased waiting delay, thus contributing to higher end-to-end delays. In any case, the system fails to meet the QoS demands for VoIP traffic.

Ideally, such real-time SUs must keep the sensing period to a minimum for ensuring the QoS of VoIP calls. However, this will, in turn, lead to increased interference (and subsequent packet loss) with PU transmissions as the probability of PU detection decreases under short sensing intervals. Therefore, in order to achieve an optimal trade-off between PU protection and QoS guarantees for SU, the basic CR cycle must be modified taking into account the relevant factors involving both VoIP applications and CRN. Additionally, it must be realized that not every channel is suitable for initiating VoIP transmission owing to different busy/idle characteristics. Hence, solutions must also be devised to detect the optimal transmission time required for VoIP applications and accordingly select the suitable channel before hosting VoIP session in a particular channel.o

All these factors lay the foundation for the proposed research study in this chapter. In this regard, a detailed literature survey is carried out in the following section that signifies the importance attached by the contemporary researchers towards successfully configuring the CR cycle parameters for achieving a balanced system involving interference-free PU transmissions and maximum throughput for SU traffic.

4.1.1 Literature Survey

Several research studies have focused on the trade-off between PU detection and channel utilization by the SU in a CR cycle. In this regard, some works have analyzed the problem of configuring the detection time (or sensing time) under the constraint of ensuring maximum throughput for the SUs. For example, a novel channel selection scheme is proposed in [4.7] that aims to maximize channel efficiency through configuration of the detection timing using analytical framework. Channel efficiency is also considered in [4.8] that proposes a numerical optimization algorithm to select the detection time and also includes the concept of multi-user detection to reduce detection probability and increase the channel utilization. Focus is shifted from the achievable data rate in these works to the actual data bits received per second that is used as the primary metric in [4.1] to obtain the sensing time and solve the sensing-throughput trade-off problem.

Since sensing and transmission intervals are inter-related, few works in literature have also studied the problem of configuring the frame slots for SU transmission. A novel scheme is proposed in [4.2] to obtain the optimal duration of data transmission in each frame. The aim is to maximize spectrum utilization while providing protection to PU traffic from interference. However, a fixed traffic model is considered for the SUs. Another work in [4.9] models the SU traffic as a discrete-time queuing model subject to bursty preemption and obtains the optimal transmission slot duration under random PU presence with the objective of maximizing throughput while keeping the Primary Interfered Time Ratio (PITR) below a tolerable limit. Traffic pattern of PUs, on the other hand, is considered in [4.3] along with the sensing time to formulate the collision-throughput problem and calculate the optimal value of frame duration to maximize SU throughput.

Few works have also been reported on the outage probability calculation in case of SUs. In this regard, an exact closed form expression is derived for the overall outage probability in [4.4] that accounts for both the probability of spectrum hole detected and the probability of channel outage for cognitive transmissions. Numerical results confirm that in order to maximize the spectrum hole efficiency, a tradeoff is indeed required in determining the time durations for spectrum hole detection and data transmission phases. This work is further extended in [4.10] which applies the CR relay scheme. Here, a cognitive relay is used for both sensing and transmission phases, based on which an exact closed-form expression is obtained, that quantifies the percentage of spectrum holes utilized by the SUs for successful data transmission. Finally, asymptotic outage analysis is conducted in [4.5] under the conditions of high SNR based on which an optimal spectrum sensing duration is obtained. This work also uses cognitive relay to improve overall outage probability.

Some other novel strategies have been reported in literature those deal with CR cycle parameters. For example, the CR users in [4.11] perform sensing and transmission at the same time through the design of a novel receiver and frame structure and in doing so, maximize both the sensing accuracy and user throughput. Another work in [4.12] provides an adaptive sensing and transmission scheme where an utility function is formulated based on three possible actions by the SU namely, i) stay idle, ii) sense and iii) transmit operations. The idea is to maximize this utility function taking into account SU throughput, PU activity detection, reliability, etc.

It is thus evident from the literature survey that there is an enormous demand for successful configuration of the CR cycle parameters. Obviously, any erroneous selection of detection and transmission cycle will adversely affect other spectrum management policies related to MAC protocols, handoff strategies, reservation schemes etc. However, the existing works in literature have not yet viewed this problem from the applications perspective. To be specific, dealing with real-time applications such as VoIP, these works require further evaluation to determine their applicability in providing QoS guarantees to the SU traffic. In addition, only limited works as in [4.12] have performed adaptive variation of sensing and transmission cycles which is a necessity while dealing with real-time transmissions. However, the main concern in these works [4.12] is the "stay idle" operation of SU, which is not feasible during VoIP communication.

Accordingly, the motivation of the work in this chapter is triggered by two factors. Firstly, the basic CRN cycle fails to provide the required trade-off between PU detection and SU throughput. Secondly, the QoS requirements of VoIP applications introduce additional constraints towards selecting the cycle parameters. Both these factors are illustrated in Fig. 4.1.

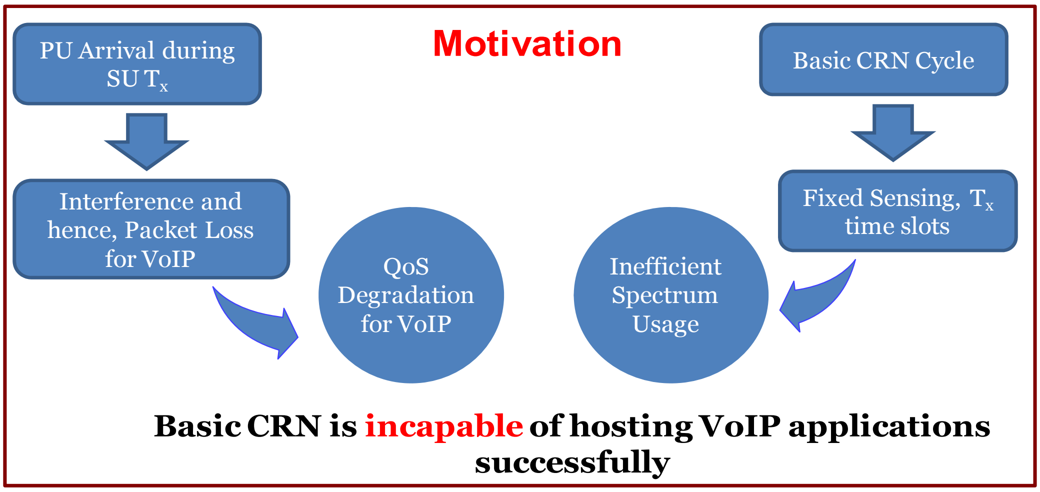


Fig. 4.1 Motivation for the Design of QoS aware CR Cycle Algorithm

4.1.2 Significant Contributions

Therefore, the objective of this chapter is to perform optimal configuration of the CR cycle parameters that include the sensing time and the transmission time. The aim is to obtain the sensing-throughput tradeoff that involves providing protection to both the PU traffic and QoS guaranteed VoIP traffic for the SUs. In this regard, the significant contributions of this chapter are explained as follows.

1. An extensive literature survey is carried out in Section 4.1 that discusses the relevance of the research problem addressed in this chapter with respect to the configuration of CR cycle parameters. It justifies the significance of carrying out research in this aspect, especially when dealing with QoS sensitive VoIP traffic. In this relation, the novelty of the work in this chapter is highlighted.

2. Optimal configuration of the sensing and transmission durations is performed in Section 4.2 by designing a suitable algorithm that is developed in two parts. In the first part, momentary sensing slots are incorporated in the transmission interval. This is followed by the adaptive variation of sensing and transmission slots based on two feedback parameters in the second part of the algorithm.

3. Comparative performance evaluation of the proposed algorithm is done in a phase-wise manner in Section 4.3. Initially, the first part of the algorithm is implemented in simulation model and analyzed both for single-channel

and multiple-channel based SU terminals. Thereafter, the second part of the algorithm is extensively analyzed under different conditions of PU traffic activity.

4. A detailed mathematical formulation is carried out in Section 4.4 that evaluates the designed algorithm and obtains critical system metrics including the total transmission time available for SUs, probability of collision, probabilities of PU detection in the sensing time and the transmission time, etc.

5. This is followed by the design of another algorithm in Section 4.5 that selects the optimal transmission time before initiating VoIP call in a particular channel. The proposed algorithm is thoroughly analyzed for performance efficiency in analytical and simulation models and also compared with the basic CR timing cycle.

6. Finally in Section 4.6, a QoS metric is defined specifically related to VoIP applications over CRN, that takes into account both the call quality metric (in terms of R-Factor) and the channel occupancy percentage. This metric is then applied to measure the performance superiority of the developed CR cycle algorithm through the generation of statistically significant data, indicating increase in both the call quality and channel occupancy percentage.

To the best of our knowledge, no such work covering all these aspects related to the configuration of CR cycle parameters for QoS aware VoIP communication by SU has been reported so far.

4.2 Proposed Algorithm for Optimal CR Timing Cycle

It is observed from the previous chapter that the sensing and transmission intervals of the SUs adversely affect the overall QoS of the VoIP calls. The basic CR timing cycle in the simulation model is therefore modified for enhancing the call quality. This modification is done in two parts. In the first part, the dedicated sensing interval is reduced and momentary sensing slots are introduced during the transmission phase of the SUs. This is followed by the adaptive variation of sensing and transmission intervals in the second part of the algorithm. Both these modules are described in this section.

4.2.1 1st Part: Introducing Momentary Sensing Slot

The first part of the designed algorithm comprises of two steps. At first, the sensing period is reduced and the transmission interval is increased to provide maximum transmission time to the VoIP SUs. However, as sensing period is decreased, incoming PU traffic may go undetected resulting in increased interference with the PU transmissions. Therefore, in the second step, this algorithm introduces momentary sensing slots where after each packet is sent, the SU momentarily senses the channel for any PU activity. On detecting the channel idle, it resumes transmission in the ongoing transmission cycle.

The network layer architecture for this modified SU node is similar to that of the OPNET model in Chapter 3 with a minor difference on the interaction between the sense node and the transmitter which is described in this section. In the previous scenario, each VoIP packet after successful transmission triggers an interrupt from the transmitter to the sense module to send the next packet till the transmission period is over. However, in the proposed algorithm, during the SU transmission period, after every packet is sent, the sense module momentarily senses the medium for any PU activity before the next transmission. The process model is henceforth modified for the sensing node of the SU terminal.

Moreover, based on the available PU traffic distribution pattern (for example, as mentioned in [4.13], [4.14]), it is possible to send more than a single packet in the transmission period before momentarily sensing the channel for the presence of PU. If the status of the PUs can be predicted to change slowly, sensing frequency (that is, how often cognitive radio should perform spectrum sensing) requirements can be relaxed [4.6]. To cite an example, the presence of a TV station usually does not change frequently in a geographical area unless a new station starts broadcasting or an existing station goes offline. At the same time, the number of packets from the SU to be sent in one transmission slot is totally dependent on the PU traffic characteristics. Thus, it implies that a trade-off is required to decide the number of packets that can be transmitted on the go before each momentary sensing phase.

Thereafter, as a possible enhancement to the proposed approach, multiple packets are sent in one transmission slot before sensing the channel momentarily in the secondary transmission interval. Also, a low bit rate codec (such as iLBC - Internet Low Bitrate Codec [4.15]) is used. In doing so, suitable modifications are made in the corresponding process model while keeping the node model unaltered for the OPNET based SU terminal. The total number of packets sent at a single transmission slot is increased from 1 to 100 and the algorithm is executed.

The overall approach is described in the flowchart as shown in Fig. 4.2.

4.2.2 2nd Part: Adaptive Variation of Timing Intervals

The first part of the algorithm aims at reducing the sensing time to a fixed value and increasing the SU transmission time with momentary sensing slots in the transmission duration, while considering fixed sensing and transmission intervals. In the second module, we eliminate this assumption of having fixed sensing times which has its own drawbacks [4.12]. Firstly, sensing the entire target spectrum continuously may be inefficient considering the power requirements and energy efficiency of the system. Secondly, the channel occupancy state can change rapidly such that the sensing mechanism may fail to keep track of the instantaneous states due to limitations on the sampling time resolution [4.16].

Therefore, an adaptive strategy is proposed where the sensing and transmission durations are varied according to the channel conditions. A feedback mechanism is deployed that aids in decision making towards runtime configuration of sensing and transmission durations. The feedback process can be implemented with the help of Sender Reports (SR) and Receiver Reports (RR) that are generated during VoIP sessions [4.17].

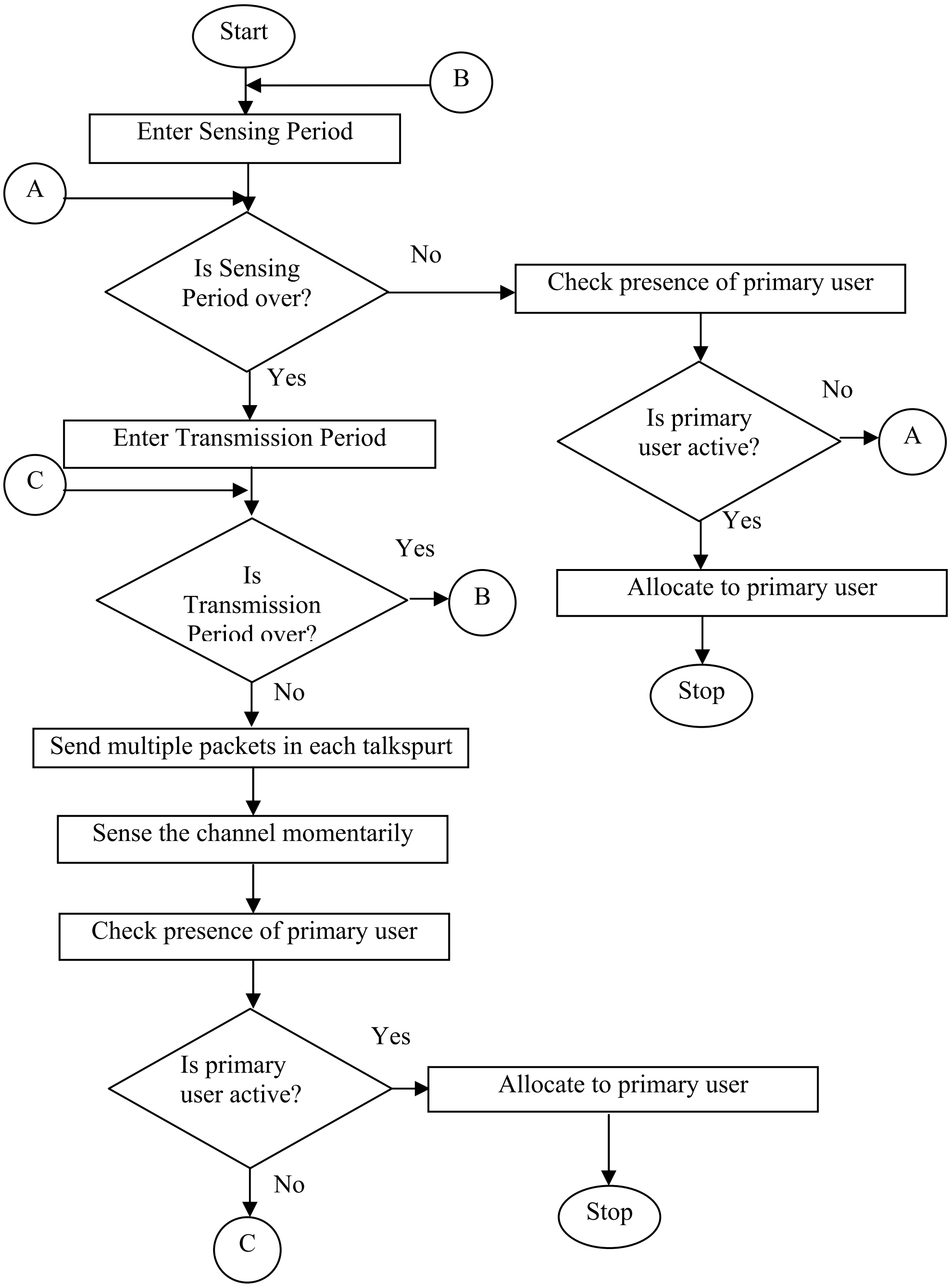


Fig. 4.2 Flowchart depicting the 1st Part of the Proposed Algorithm

The adaptive variation of timing intervals which forms the second part of the algorithm is henceforth described as follows.

Adaptive Variation of Timing Intervals	
Step 1:	Start VoIP transmission with a high sensing time (*ts*) and low secondary transmission time (*td*).
Step 2:	Set i=1. Start the i^{th} cognitive cycle.

Step 3:	Check for any interference with the PU in $t_{d\,i-1}$ duration. Let it be I_{check}.
Step 4:	If I_{check}=true, $t_{si} = t_{si} + K_1$, $t_{di} = t_{di} - K_1$.
Step 5:	If I_{check} =false, $t_{si} = t_{si} - K_2$, $t_{di} = t_{di} + K_2$.
Step 6:	If VoIP transmission is not over, $i = i+1$, goto *Step 7* else goto *Step 8*.
Step 7:	Goto *Step 2*.
Step 8:	Calculate VoIP QoS metrics and CRN parameters.

Both these modules are pictorially represented in the form of a schematic diagram in Fig. 4.3.

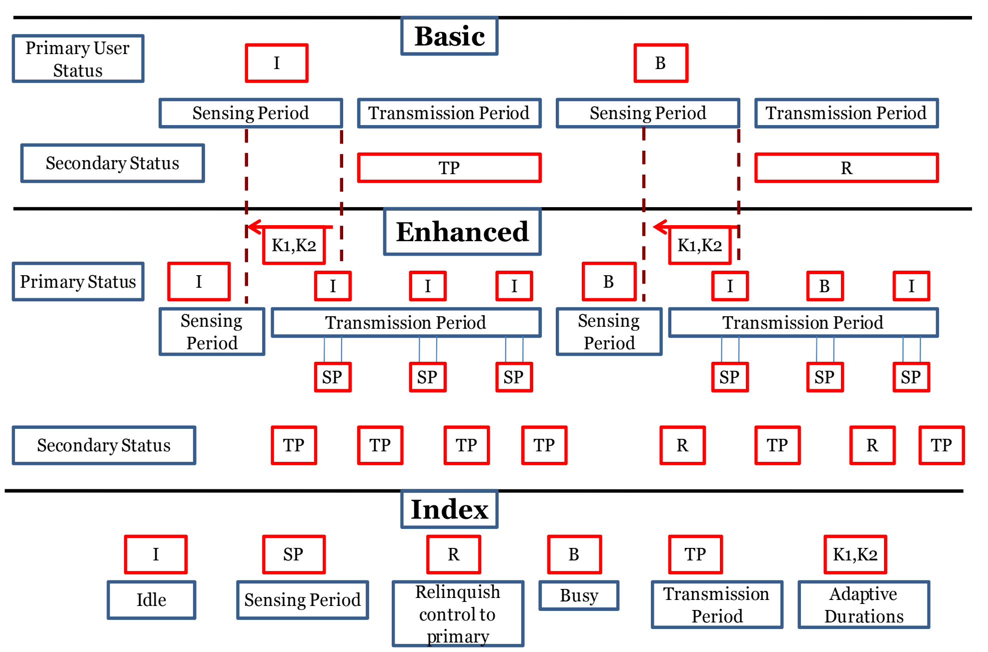


Fig. 4.3 Schematic Diagram illustrating the Proposed Algorithm in a nutshell

4.3 Comparative Performance Evaluation of the Proposed Algorithm

The proposed algorithm is now analyzed using the simulation model which has been developed in OPNET Modeler 16.0.A (described in Chapter 2) and suitably modified to incorporate all the aspects of this algorithm. The analysis is carried out in two phases. Initially, the effect of incorporating the momentary sensing slots is studied which constitutes the first part of the algorithm. Thereafter, VoIP performance is analyzed under adaptive variation of sensing and transmission durations, which form the second part of the designed algorithm.

4.3.1 Analysis of the 1st Part: Momentary Sensing Slots

(i) OPNET Model output under Single Channel Scenario

It is observed from Fig. 4.4 and Fig. 4.5 respectively that the end-to-end delay and packet loss are reduced after implementation of the proposed modification in the CR timing cycle. The graphs are plotted with increasing sensing intervals for each transmission period.

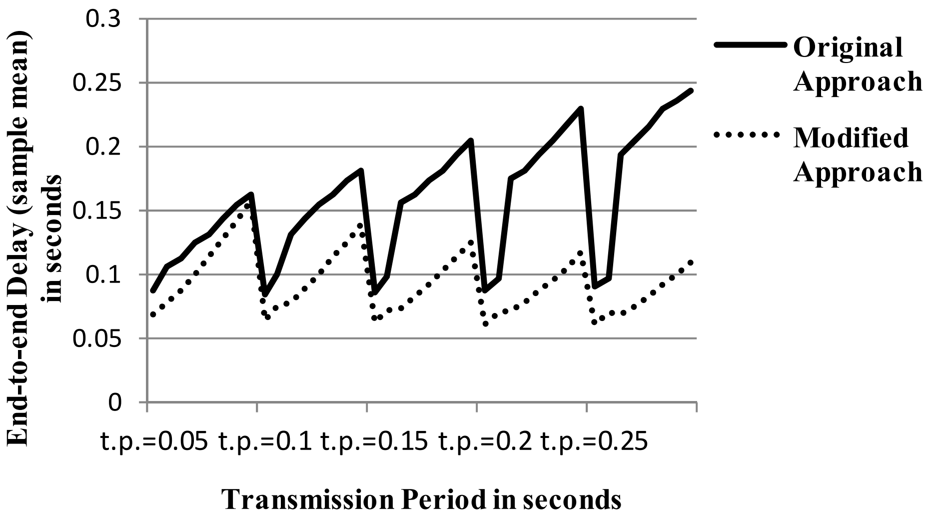


Fig. 4.4 Variation in end-to-end delay (sample mean) for the original and the modified approach

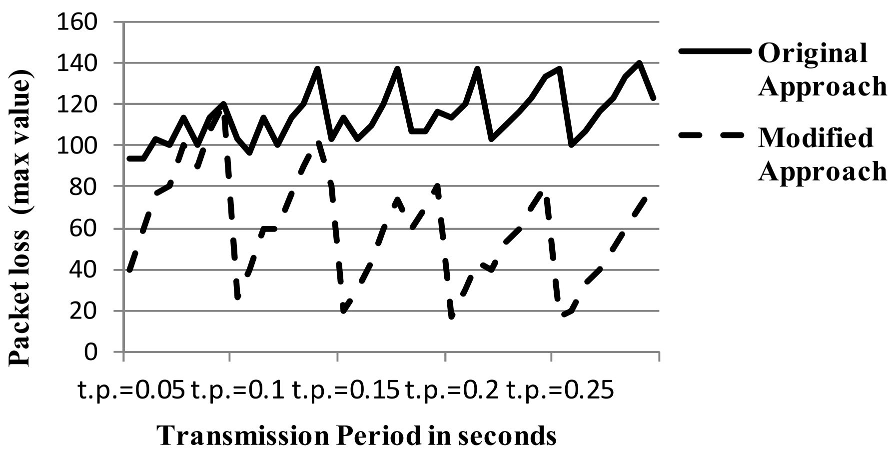


Fig. 4.5 Variation in packet loss (maximum value) for the original and the modified approach

Further, it is observed from Fig. 4.6 that the standard deviation of the total received SU traffic from the mean value is reduced in this scenario. This subsequently reduces the jitter as reflected in Fig. 4.7. Simulation readings illustrate that the delay remains within the threshold limit of 150 ms while jitter gradually decreases below 100 ms which is acceptable with respect to voice call.

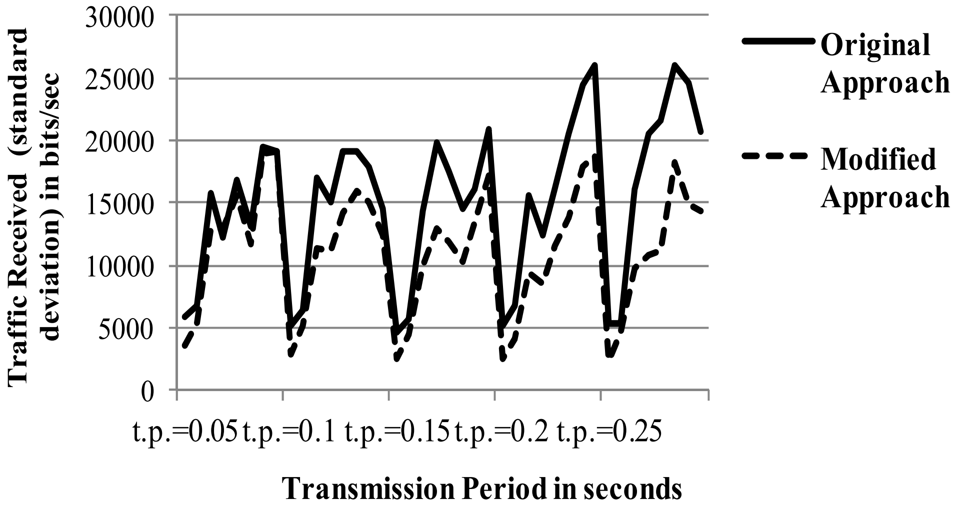


Fig. 4.6 Variation in traffic received (standard deviation) under the original and the modified approach

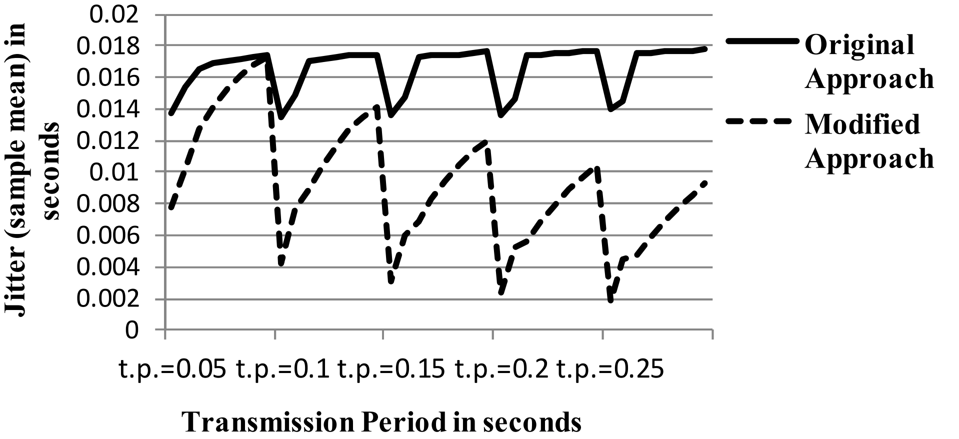


Fig. 4.7 Variation in jitter (sample mean) for the original and the modified approach

Finally, as observed from Fig. 4.8, further enhancement to the proposed approach as suggested in Section 4.2.1 reduces packet loss to a minimum. While the PU is inactive, more number of packets is sent in one transmission slot before momentary sensing of the channel during the SU transmission period. Therefore, packet loss occurs only during the sensing intervals where more packets contend for the medium.

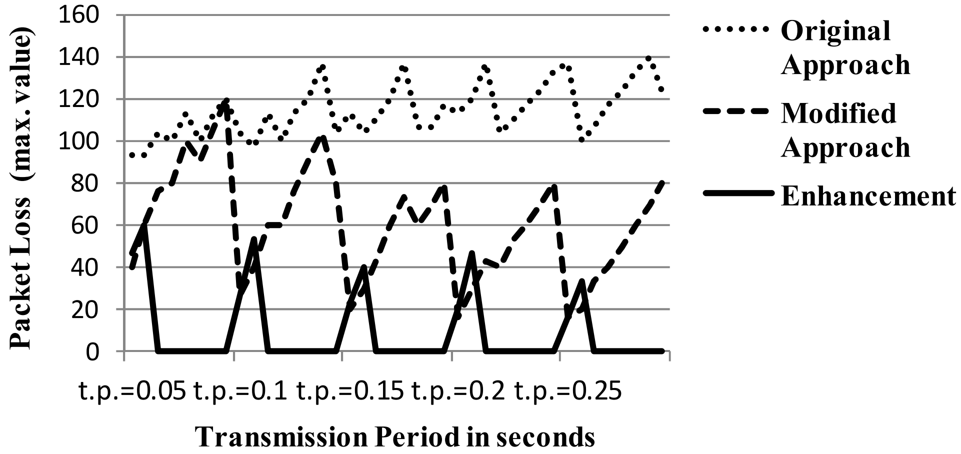


Fig. 4.8 Packet loss (maximum value) for the basic CR timing cycle, Modified CR timing cycle and finally the enhanced CR timing cycle

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

(ii) OPNET Model output under Multiple Channel Scenario

In the next phase, the multiple-channel scenario based OPNET model (as described in Chapter 3) is used to analyze VoIP performance in CRN following the basic CR principle and the proposed algorithm (as mentioned in Section 4.2.1). It is clearly observed in Fig. 4.9 that overall increase in end-to-end delay during VoIP communication is much less in the enhanced scenario as compared to the basic scenario. It is to be noted that there is a delay associated with switching to the other channel whenever the current channel is sensed busy. This operation involves active participation and decision making with respect to the *MAC_Controller* node of the SU terminal and further contributes to the overall spectrum handoff delay [4.18]. Under low PU activity, the SU can still perform this handoff due to imperfect spectrum sensing, especially during the false alarm scenarios. In the basic CRN, the number of such occurrences with respect to channel switching is more, thereby increasing the delay. As the switching of channels takes place only in the sensing time, the total number of occurrences with respect to channel switching decreases with reduction in sensing time in the proposed algorithm, resulting in a drastic reduction in delay.

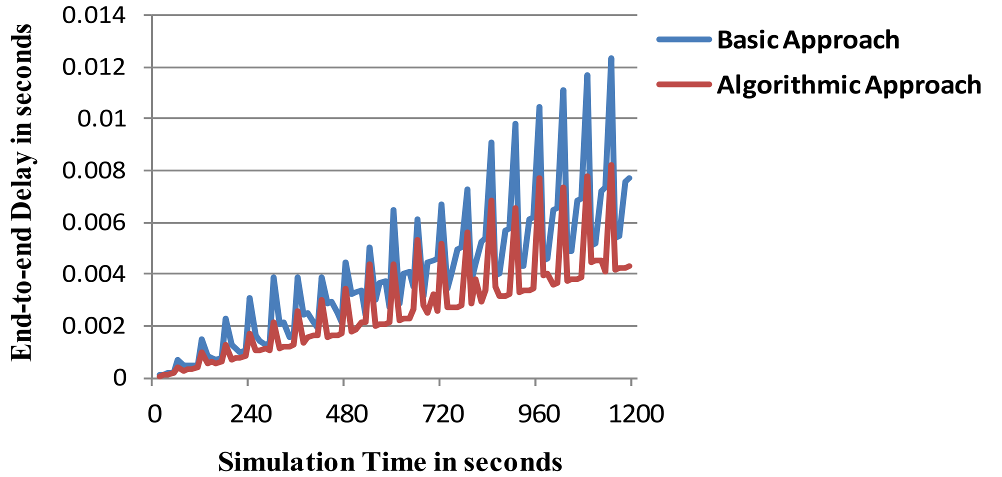


Fig. 4.9 Variation in end-to-end delay for an ongoing VoIP session for the basic and the algorithmic approach

Moreover, each time a new channel is marked ready for SU transmission, the spectrum handoff probability increases with time which results in the handoff delay as already described. This delay is attributed as a variable component to the total delay incurred during VoIP transmissions. Therefore, in the basic CRN scenario, total number of variable delay components increases than in the enhanced scenario. As jitter [4.17] is the difference between successive delays, increase in these variable delay components increases jitter. Hence the basic CR timing cycle witnesses higher values of jitter as compared to the proposed algorithm and the same is recorded in Fig. 4.10.

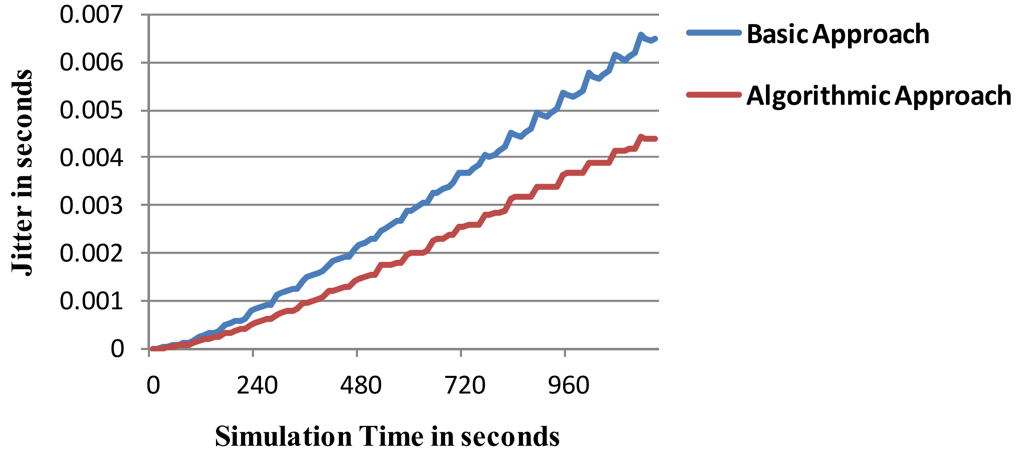


Fig. 4.10 Variation in jitter for an ongoing VoIP session for the basic and the algorithmic approach

Variation in the rate of received packets provides a clear idea of the packet loss in the model. Fig. 4.11 shows the throughput with respect to SU receiver. It is observed that the overall throughput decreases in the basic CRN scenario as compared to the algorithmic scenario.

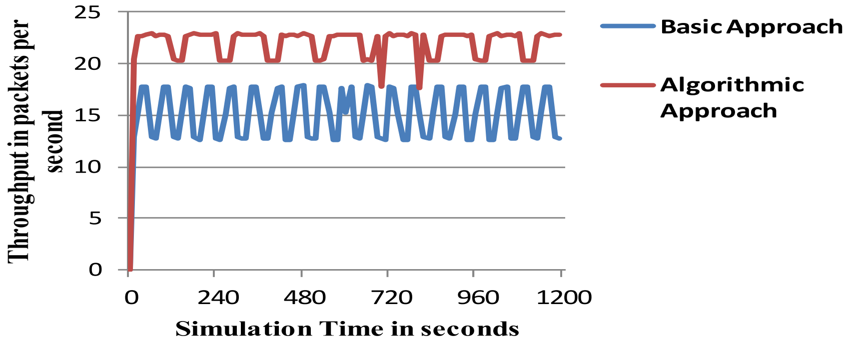


Fig. 4.11 Variation in throughput for the basic and the algorithmic approach

4.3.2 Analysis of the 2nd Part: Adaptive Variation of Timing Intervals

The second part of the proposed algorithm is implemented by suitable modifications to the developed model as already described in Section 4.2.2. Selection of ts, td, K_1 and K_2 are crucial for optimum performance of VoIP in CRN for a particular scenario and hence must be chosen carefully using an analytical framework (discussed in the next section). We have chosen two scenarios for low and high PU activity corresponding to the average and worst scenarios respectively to analyze VoIP performance under the basic CR timing cycle and the adaptive CR timing cycle based on the designed algorithm. It is to be noted that the basic CR timing cycle is denoted by "Non-Adaptive scenario" in this section to highlight its difference with the designed "Adaptive Approach".

(i) Simulation Output under Low PU activity

In the scenario where the frequency of PU arrival is less, both the non-adaptive and adaptive approaches are implemented. Here the PU arrival follows a Poisson distribution with the mean on-time of 10 ms and the mean off-time of 500 ms. It is observed from Fig. 4.12 that the throughput of the channel (where the PU arrives intermittently) with respect to the secondary VoIP user is higher when the adaptive strategy is implemented. On the other hand, the throughput remains at an average constant value for the non-adaptive strategy even for low PU activity. This is due to the inability of the basic CR timing cycle to take advantage of idle channel conditions during low PU traffic.

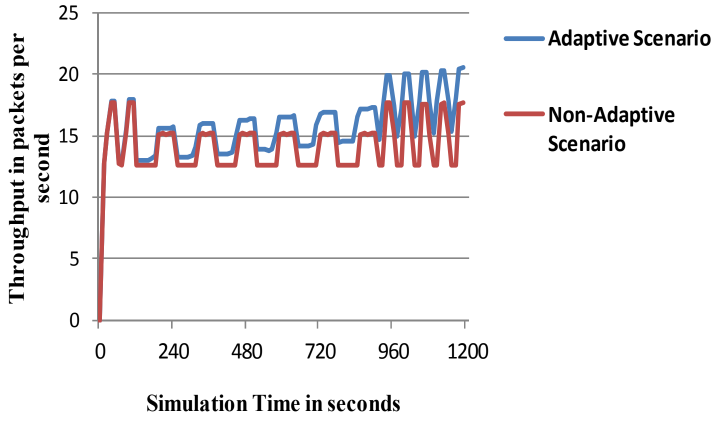


Fig. 4.12 Variation in throughput for adaptive and non-adaptive scenarios during low PU activity

The variation of end-to-end delay is depicted in Fig. 4.13. It is observed that the increase in delay for the adaptive algorithm is less than that of the non-adaptive scenario. The reason is clearly embedded in the fact that adaptive principle minimizes the time wasted in channel sensing and hence, the associated time delay is minimized.

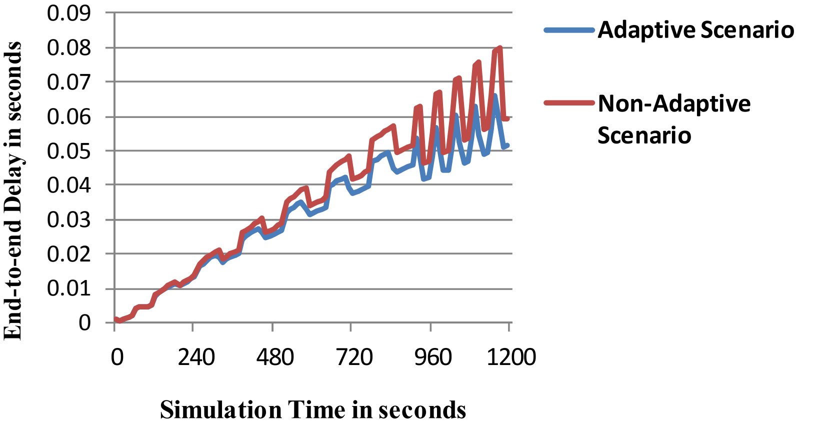


Fig. 4.13 Variation in end-to-end delay for adaptive and non-adaptive scenarios during low PU activity

Further, the increase in jitter is reduced after implementation of the adaptive algorithm as reflected in Fig. 4.14. This is because as the alternate sensing and transmission period switching is minimized, the variable delay component (that contributes to jitter) is also reduced under the proposed adaptive approach.

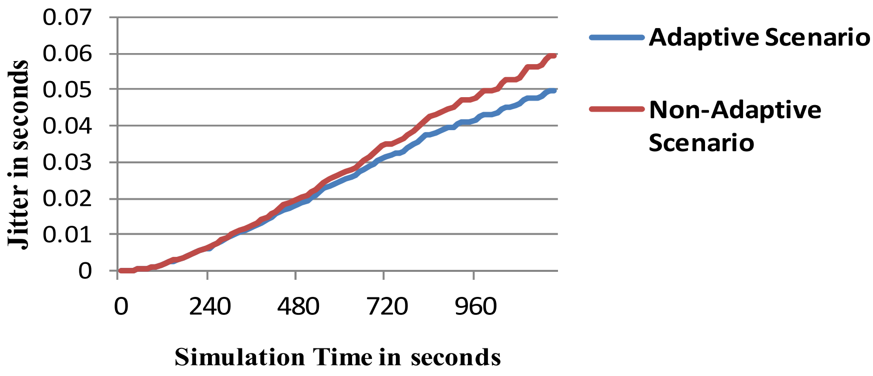


Fig. 4.14 Variation in jitter during an ongoing VoIP session for adaptive and non-adaptive scenarios during low PU activity

(ii) <u>Simulation Output under High PU activity</u>

The adaptive approach is further analyzed for performance improvement in worst case scenarios having high PU activity. Unlike the previous scenario, here the PU arrival follows a Poisson distribution with the mean on-time of 100 ms and the mean off-time of 500 ms. Fig. 4.15 reflects the throughput of an SU implementing VoIP calls. It is seen that the adaptive approach has a similar performance compared to the non-adaptive algorithm even during high PU activity. Obviously, the mean throughput for both adaptive and non-adaptive approaches falls during the on-off activities of the PU traffic, and increases when the channel is again free from any PU presence. Overall, it fares slightly better than the non-adaptive strategy as far as the throughput is concerned. Therefore, the channel utilization also rises as depicted in Fig. 4.16 where the adaptive technique performs better compared to the other one.

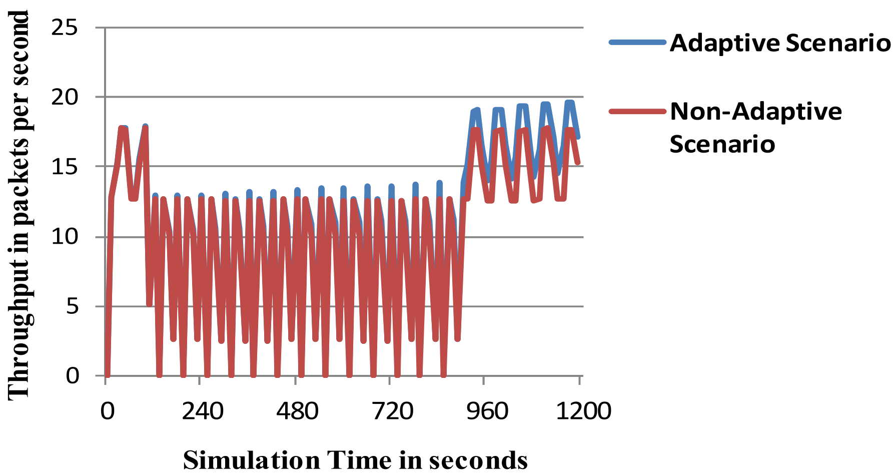


Fig. 4.15 Variation in throughput of SUs for adaptive and non-adaptive scenarios during high PU activity

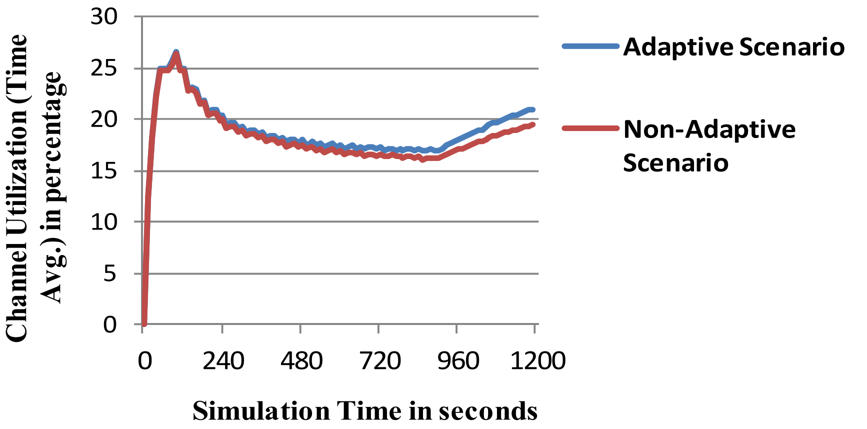


Fig. 4.16 Channel utilization (Time Avg.) under adaptive and non-adaptive scenarios during high PU activity

Also, the analysis of delay and jitter from Fig. 4.17 and Fig. 4.18 respectively points to the efficiency of the adaptive strategy as it helps in restricting the values of delay and jitter within 100 ms and 80 ms respectively (within their threshold limits), clearly reflecting the high quality of VoIP call.

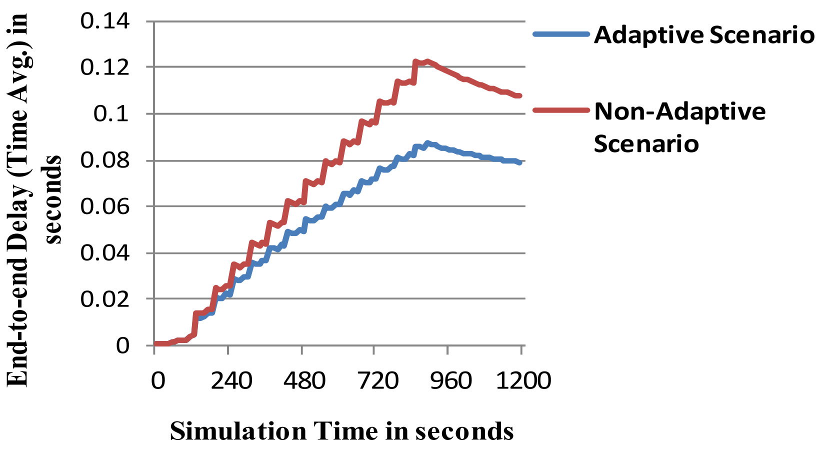


Fig. 4.17 Variation in end-to-end delay (time avg.) under adaptive and non-adaptive scenarios during high PU activity

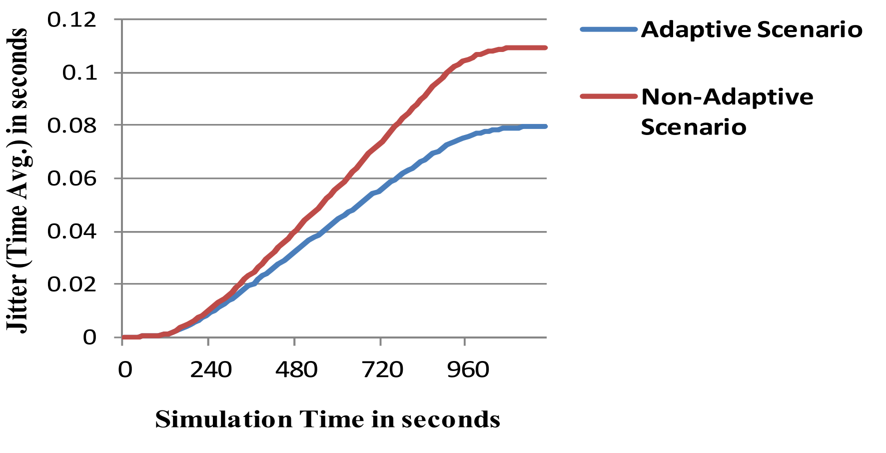


Fig. 4.18 Variation in jitter (time avg.) for an ongoing VoIP session under adaptive and non-adaptive scenarios during high PU activity

Implementation results visibly indicate the efficiency of the adaptive algorithm in enhancing the required call quality, thereby validating its performance superiority compared to the basic CR timing cycle.

Thus, it is inferred from this section that both the algorithmic modules significantly contribute towards sustaining long duration VoIP communication, which was not possible under the purview of the basic CR timing cycle. The next step is to analyze the merits of the algorithm using analytical formulation and is discussed as follows.

4.4 Mathematical Formulation

This section provides a basic mathematical overview of the proposed approach in this chapter and validates the findings from the simulation results as obtained in OPNET Modeler 16.0.A. An important observation from the previous sections is that careful configuration of the different timing and feedback parameters is essential towards realizing the benefits of the design methodology. Accordingly, the mathematical formulation in this section derives the necessary conditions that in turn will lead to proper configuration of the related parameters (t_d, t_p, t_r, K_1, K_2).

Initially, the representation of the basic CR timing cycle along with the1st and 2nd parts of the designed algorithm are illustrated in Fig 4.19 (a), (b) and (c) respectively for better understandability.

Accordingly, the expressions for the total time of CR cycle are derived corresponding to Fig 4.19. Let t_{cog}, t_{si} and t_{di} be the time equivalent to the total CR timing cycle, sensing time and transmission time in the i^{th} CR timing cycle respectively. Let P_{psi} denote the probability of PU detection in the i^{th} sensing time (denoted by t_{si}) and let P_{ctdi} be the probability of detecting collision with the PU in the i^{th} transmission time (denoted by t_d). Therefore, the total time $tcog$ is given by the following expression.

$tcog = \sum_{over\ all\ cycles}$ [for each cycle "i": {(*sensing time* + K_1) if PU presence is detected in the previous transmission cycle else (*sensing time* – K_2)} +{given that the channel is detected idle in the previous sensing time, (*transmission time*

- K_1) if PU presence is detected in the previous transmission cycle else (*transmission time* + K_2)}]

$$
\begin{aligned}
\Rightarrow tcog = & \left\{t_{s1} + t_{d1} \times \left(1 - P_{ps1}\right)\right\} \quad \leftarrow \text{1}^{st}\text{ cycle} \\
& \quad \text{2}^{nd}\text{ cycle} \\
& + \left\{\left(t_{s2} + K_1\right) \times P_{ctd1} + \left(t_{s2} - K_2\right) \times \left(1 - P_{ctd1}\right)\right\} \\
& + \left(1 - P_{ps2}\right) \times \left\{\left(t_{d2} - K_1\right) \times P_{ctd1} + \left(t_{d2} + K_2\right) \times \left(1 - P_{ctd1}\right)\right\} \\
& \quad \text{n}^{th}\text{ cycle} \\
& \ldots\ldots + \left\{\left(t_{sn} + K_1\right) \times P_{ctdn-1} + \left(t_{sn} - K_2\right) \times \left(1 - P_{ctdn-1}\right)\right\} \\
& + \left(1 - P_{psn}\right) \times \left\{\left(t_{dn} - K_1\right) \times P_{ctdn-1} + \left(t_{dn} + K_2\right) \times \left(1 - P_{ctdn-1}\right)\right\}
\end{aligned}
\qquad (4.1)
$$

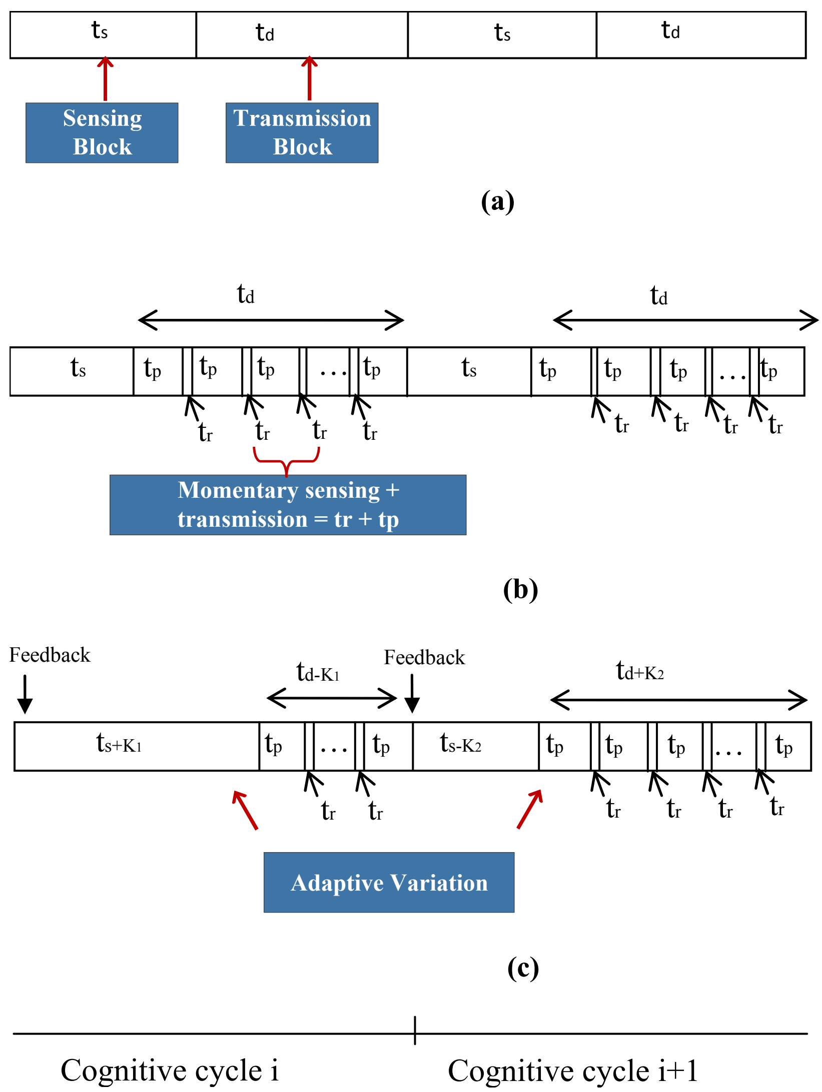


Fig. 4.19 Representation of (a) basic, (b) modified non-adaptive and (c) adaptive cognitive radio cycle

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

Deriving expressions for T_s and T_d by expanding (4.1),

Sensing Block

$$T_s = t_{s1} + \sum_{i=2}^{n} \left\{ t_{si} + P_{ctdi-1} \times \left(K_1 + K_2\right) - K_2 \right\} \tag{4.2}$$

Transmission Block

$$\begin{aligned} T_d = {} & \sum_{i=1}^{n} \left\{ t_{di} \times \left(1 - P_{psi}\right) \right\} \\ & + \sum_{i=2}^{n} \left[\left(1 - P_{psi}\right) \times \left\{ K_2 - P_{ctdi-1} \times \left(K_1 + K_2\right) \right\} \right] \end{aligned} \tag{4.3}$$

As the objective is to maximize t_d, both (4.2) and (4.3) are solved further to arrive at the condition as expressed in (4.4).

$$\sum_{i=2}^{n} P_{ctdi-1} \quad << \quad \frac{K_2 \times (n-1)}{K_2 + K_1} \tag{4.4}$$

Thereafter, the expression for the individual transmission period t_d as per the 1st part of the proposed algorithm (described in Section 4.2.1) is derived after incorporating the effect of the adaptive technique based on the 2nd part of the algorithm (as stated in Section 4.2.2).

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

When collision occurs in the i^{th} transmission slot,

$$\begin{aligned} t_{di} &= \left(N - K_1'\right) \times \left(t_r + t_p\right) \\ &= N_c \times \left(t_r + t_p\right) \end{aligned} \tag{4.5}$$

where $K_1 = K_1' \times \left(t_p + t_r\right)$, $N_c = N - K_1'$ (4.6)

When no collision occurs in the i^{th} transmission slot,

$$t_{d_i} = \left(N + K_2'\right) \times \left(t_r + t_p\right)$$

$$= N_{nc} \times \left(t_r + t_p\right) \qquad (4.7)$$

where $K_2 = K_2' \times \left(t_p + t_r\right)$, $N_{nc} = N + K_2'$ (4.8)

Thus, the number of (t_p+t_r) cycle increases in this case as per (4.8) due to overall increase in the transmission cycle (t_{di}) block.

Let P_{cri} = probability of PU detection in the i^{th} momentary sensing period, $P_{tpi>0}$ = probability that successful transmission occurs during i^{th} t_p slot. Accordingly, deriving the expression for each transmission block (t_d),

$t_d = \sum_{i=1}^{N}$ [for each cycle "i" within t_d : *transmission time* (if PU absence is detected in the previous momentary sensing) + *momentary sensing time* (if successful transmission occurs in the previous transmission slot)

$$\Rightarrow t_d = \left[t_{p1} + \overset{\text{1st cycle}}{\downarrow}\, t_{r1}\right] + \left[\left\{t_{p2} \times \left(1 - P_{cr1}\right)\right\} + \overset{\text{2nd cycle}}{\downarrow}\left\{t_{r2} \times \left(P_{tp2>0}\right)\right\}\right] + \ldots\ldots$$
$$+ \left[\left\{t_{pj} \times \left(1 - P_{crj-1}\right)\right\} + \left\{t_{rj} \times \left(P_{tpj>0}\right)\right\}\right] + \ldots\ldots \longleftarrow \text{j}^{\text{th}}\text{ cycle}$$
$$+ \left[\left\{t_{pN-1} \times \left(1 - P_{crN-2}\right)\right\} + \left\{t_{rN-1} \times \left(P_{tpN-1>0}\right)\right\}\right] \longleftarrow \text{(N-1)}^{\text{th}}\text{ cycle}$$
$$+ \left[\left(t_{pN} + t_{rN}\right) \times \left(1 - P_{crN-1}\right)\right] \longleftarrow \text{N}^{\text{th}}\text{ cycle}$$

$$= t_{p1} + t_{r1} + \sum_{j=2}^{N-1} t_{pj} \times \left(1 - P_{crj-1}\right) + \sum_{j=2}^{N-1} t_{rj} \times P_{tpj>0}$$
$$+ \left[\left(t_{pN} + t_{rN}\right) \times \left(1 - P_{crN-1}\right)\right] \qquad (4.9)$$

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

Therefore, (4.9) can be simplified accordingly and is expressed as,

$$t_d = \underbrace{t_p \times \left(1 + \sum_{j=2}^{N} \left(1 - P_{crj-1}\right)\right)}_{\text{SU Transmission in between sensing slots}} + \underbrace{t_r \times \left(2 - P_{crN-1} + \sum_{j=2}^{N-1} P_{tpj>0}\right)}_{\text{Total momentary sensing slots}} \qquad (4.10)$$

Hence, in order to maximize t_d, the following conditions must be satisfied.

$$\sum_{j=2}^{N} P_{crj-1} \quad << N \tag{4.11}$$

$$\sum_{j=2}^{N-1} \left(P_{tpj} > 0\right) - P_{crN-1} \approx N-2 \tag{4.12}$$

For collision in the (i-1)th case, (4.11) and (4.12) can be expressed as follows.

$$\sum_{j=2}^{Nc} P_{crj-1} \quad << N_c \tag{4.13}$$

$$\sum_{j=2}^{Nc-1} \left(P_{tpj} > 0\right) - P_{crNc-1} \quad \approx N_c - 2 \tag{4.14}$$

For no collision in the (i-1)th case, (4.11) and (4.12) can be expressed as follows.

$$\sum_{j=2}^{Nnc} P_{crj-1} \quad << N_{nc} \tag{4.15}$$

$$\sum_{j=2}^{Nnc-1} \left(P_{tpj} > 0\right) - P_{crNc-1} \quad \approx N_{nc} - 2 \tag{4.16}$$

Expanding (4.4), we get

$$P_{ctd1} + P_{ctd2} + P_{ctd3} + \ldots\ldots P_{ctdn-1} \quad << \frac{K_2 \times (n-1)}{K_1 + K_2} \tag{4.17}$$

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

Therefore, (4.17) is modified accordingly, that leads to the following condition.

$$f_1\left(P_{cr1}, P_{cr2}, \ldots P_{crN-1}\right) + f_2\left(P_{cr1}, P_{cr2}, \ldots P_{crg2(f1)}\right) + f_3\left(P_{cr1}, P_{cr2}, \ldots P_{crg3(f2)}\right) + \ldots + f_{n-1}\left(P_{cr1}, P_{cr2}, \ldots P_{crgn-1(fn-2)}\right) << \frac{K_2 \times (n-1)}{K_1 + K_2}$$

$$\Rightarrow f_1\left(P_{cr1}, P_{cr2}, \ldots P_{crN-1}\right) + \sum_{l=2}^{n-1} f_l\left(\left(P_{cr1}, P_{cr2}, \ldots P_{crgl(fl-1)}\right)\right) << \frac{K_2 \times (n-1)}{K_1 + K_2} \tag{4.18}$$

where $g_l(f_l\text{-}1)$ is a functional relationship whose value is an integer and is expressed as follows.

$$g_l(f_l - 1) = N_c \text{ for collision in the previous } t_s + t_d \text{ cycle.} \tag{4.19}$$

$$g_l(f_l - 1) = N_{nc} \text{ in absence of collision.} \tag{4.20}$$

Therefore, from (4.18), it is observed that the condition for maximum transmission duration depends on P_{cri} which in turn depends on (4.13), (4.14), (4.15) and (4.16). It is quite clear from (4.18) that the optimal selection of K_1 and K_2 is very crucial towards the proper functioning of the modified CR timing cycle and can be configured using the derived conditions.

4.5 VoIP Call Initiation: Selection of Optimal Transmission Duration

After configuring the timing intervals, the next step is to select the optimal transmission duration before initiating the VoIP session. This is very important because VoIP calls cannot be initiated unless adequate transmission time is guaranteed for the VoIP traffic. This is particularly applicable during default channel assignment to the SU. Only when the channel is idle for a considerable amount of time can it be used by the SU to perform VoIP communication, or else it must perform spectrum handoff and select a new channel. Thus, in this section apart from detecting the channel busy/idle characteristics, we check another condition as follows.

"If the channel is idle, what is the expected idle time duration and is it worthy for supporting QoS sensitive real-time calls?"

Initially, the problem statement is defined followed by algorithm design, analysis, and implementation.

4.5.1 Problem Statement

In the basic CR timing cycle, SU has fixed sensing and transmission time intervals. The SU senses the channel during its sensing period and starts its transmission in the transmission period only when PU is inactive. With PU arrival, SU performs spectrum handoff and moves to another available idle channel. However, due to imperfect sensing, the problems of false alarm and miss-detection [4.19] occur, thereby resulting in loss of spectrum utilization and interference with PU respectively.

Let T_s and T_D be the sensing and transmission time intervals for SU within one time slot denoted by T. Let P_{FA} be the false alarm probability. Therefore, the optimal transmission time T_D' is derived as per [4.8] and expressed in (4.21).

$$\begin{aligned} T_D' &= T_D(1-P_{FA}) \\ &= (T-T_S)(1-P_{FA}) \end{aligned} \tag{4.21}$$

Let α and β be the transition rates from busy to idle and from idle to busy periods relating to the channels respectively. The probability distribution functions of channel busy and idle periods are denoted by f_{BR} and f_{IR} respectively.

Considering exponential distribution [4.20], f_{BR} and f_{IR} are expressed in (4.22) and (4.23) respectively as per [4.7].

$$f_{BR}(t) = \alpha e^{-\alpha t} \tag{4.22}$$

$$f_{IR}(t) = \beta e^{-\beta t} \tag{4.23}$$

The effective and non-effective communication durations in case of normal scenario (not false alarm) and miss-detection scenario are denoted by $T_{I,C}$ and $T_{B,P}$ respectively and their corresponding expressions are derived as follows [4.7]. It is considered that the channel changes its state (as detected to be idle by the SU) after t_I time period.

$$T_{I,C} = \int_{T_D'}^{\infty} f_{IR}(t_1)\, T_D'\, dt_1 + \int_{0}^{T_D'} f_{IR}(t_1)\, t_1 dt_1 \tag{4.24}$$

$$T_{B,P} = \int_{T_D'}^{\infty} f_{BR}(t_1)\, T_D'\, dt_1 + \int_{0}^{T_D'} f_{BR}(t_1)\, t_1 dt_1 \tag{4.25}$$

However, this scenario is not suitable for real-time VoIP communication. This is because with PU arrival and subsequent SU spectrum handoff, the overall delay incurred and the interference with PU during SU transmission adversely affect the QoS of VoIP calls. Therefore, the available transmission time must be allotted to SU for VoIP communication only after proper analysis so that the following constraints are satisfied.

1. Interference with PU is minimized.
2. Number of spectrum handoffs and hence the associated delay is reduced.

Hence, the motivation for this work is clearly highlighted in Fig. 4.20.

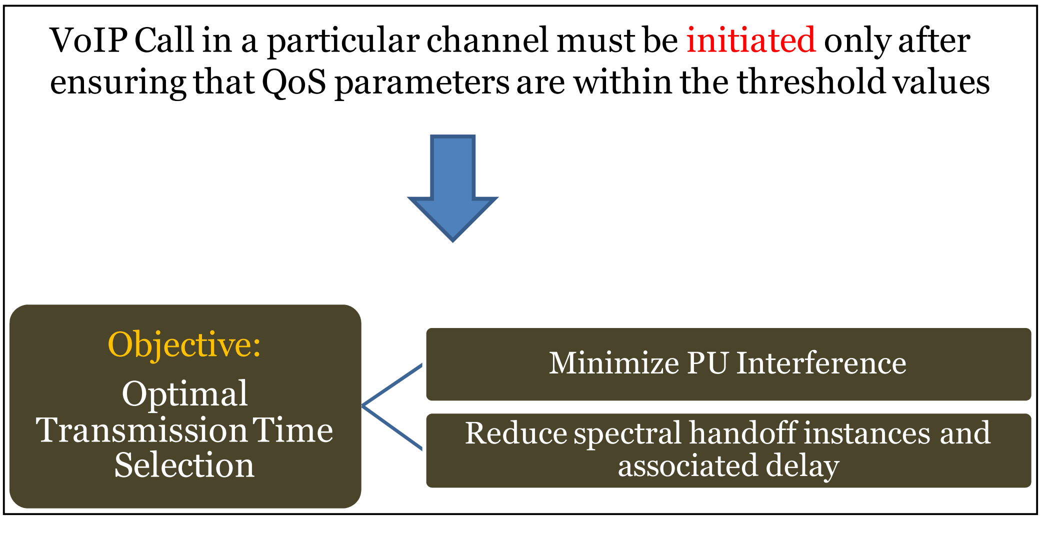


Fig. 4.20 Requirement of the Optimal Transmission Time for VoIP call initiation

4.5.2 Proposed Algorithm

The proposed algorithm aims to allocate optimal transmission time for VoIP communication and thereby satisfy the constraints as mentioned in Section 4.5.1. In this approach, three significant aspects have been considered to maintain the QoS of VoIP transmission and are mentioned as follows.

1. Possible action to guard against any miss-detection. (Selection of T'' parameter).
2. Primary User traffic prediction to avoid interference due to sudden PU arrival. (Incorporation of P_{PU} - Probability of PU arrival).
3. Selection of transmission time that is optimal for VoIP communication in SU. (Calculation of $T_{overall}$ parameter).

The algorithm is described as follows.

Proposed Algorithm: Selection of Optimal Transmission Time for VoIP call	
Step 1:	Calculate the probability of PU arrival (denoted by P_{PU}). It can be done either by considering certain traffic distribution for PU arrival or by gathering information from the past occurrences. The calculation must be done by SU controller (SC node) and the outcome should be communicated to individual SUs.
Step 2:	Calculate T'' which is initially taken as $(1/10)T_D'$. If PU interference occurs due to miss-detection in the current time slot, T'' is selected to be *max* $\{(T''_{prev}+(1/10)T_D'), (1/2)T_D'\}$. In absence of any PU interference in the present time slot, T'' is selected as *min* $\{(T''_{prev}-(1/10)T_D'), (1/10)T_D'\}$, where T''_{prev} is the value of T'' in the previous instance.
Step 3:	Calculate the proposed optimal transmission time (denoted by $T_{propfinal}$) which is expressed as

$$T_{propfinal} = (T_D - T'')(1 - P_{PU})$$

$$= \{(T - T_S)(1 - P_{FA}) - T''\}(1 - P_{PU})$$

(4.26)

Step 4: If $T_{propfinal}$ >= n * average talkspurt time period for VoIP communication (where n is an integer), $T_{overall} = T_{propfinal}$ and it is allocated to SU for VoIP transmission, else it is rejected.

The entire algorithm is depicted in the flowchart as denoted by Fig. 4.21. The reason for the initial selection of T'' is that considering the general scenario of having a small T_D time slot, any increase in T'' will result in the channel remaining idle, thereby defeating the purpose of CRN to increase spectrum utilization. On the other hand, decreasing T'' further increases the probability of miss-detection, thereby degrading the VoIP QoS. Furthermore, as T'' is adaptively varied as per PU activity, the initial selection of T'' is justified.

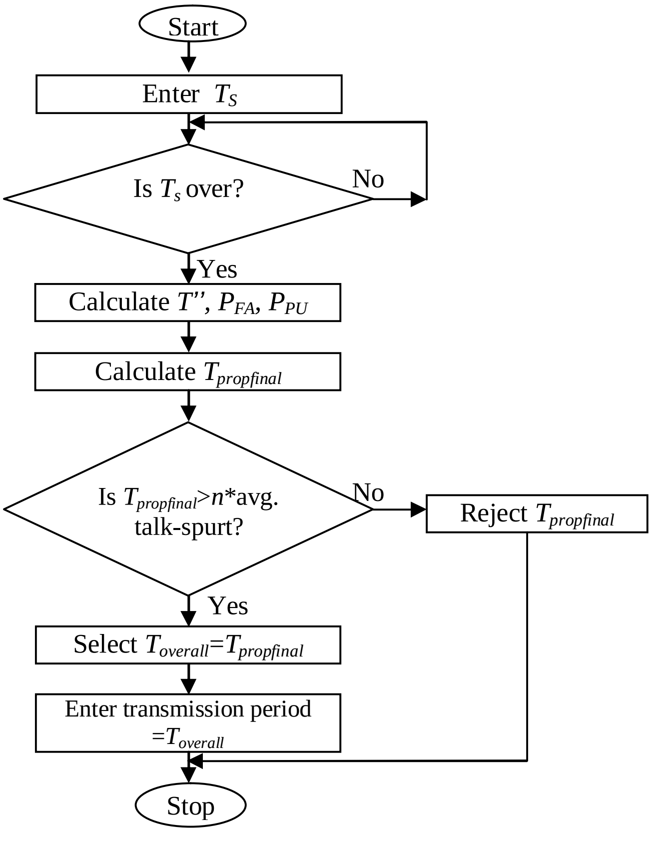


Fig. 4.21 Flowchart depicting the proposed algorithm

4.5.3 Mathematical Analysis of the Proposed Algorithm

The proposed algorithm is analyzed mathematically to evaluate its performance with respect to interference reduction and VoIP QoS maintenance. In particular, this section highlights the strengths and weaknesses of the proposed algorithm through analytical observations.

(i) ***Loss in Utilization***

It is clearly observed from (4.26) that the actual time granted for SU transmission is reduced and hence it results in a certain loss in spectrum utilization. Let $Loss_{util}$ be the percentage loss in utilization with respect to time and is expressed as follows.

$$Loss_{util} = \frac{T_D' - T_{overall}}{T_D'} \times 100 \tag{4.27}$$

Substituting T_D' and $T_{overall}$ $(=T_{propfinal})$ from (4.21) and (4.26) respectively,

$$Loss_{util} = \left[P_D + \left\{ \frac{T''}{(T - T_S)(1 - P_{FA})} (1 - P_{PU}) \right\} \right] \times 100 \tag{4.28}$$

Two cases are observed from (4.28).

Case 1: For $P_{PU} = 1$(implying definite PU arrival), we have

$Loss_{util} = 100$ %

Case 2: For $P_{PU} = 0$ (implying no PU activity), we have

$$Loss_{util} = \frac{T''}{(T - T_S)(1 - P_{FA})} \times 100 \tag{4.29}$$

It is, therefore, concluded from (4.29) that $Loss \propto T''$ which indicates that during low PU activity, T'' must be as less as possible to reduce loss in spectrum utilization. This justifies the decision to adaptively vary T'' in the proposed algorithm.

(ii) ***Loss in Effective Communication Duration***

Loss in channel utilization (as already discussed) further leads to perceivable loss in throughput in the normal (not false alarm) scenario. Let

$Loss_{th\_nor}$ be the loss in effective communication duration after implementation of the proposed algorithm and is expressed as follows.

$$Loss_{th\_nor} = \left[\int_{T_D'}^{\infty} f_{IR}(t_1) T_D' dt_1 + \int_{0}^{T_D'} f_{IR}(t_1) t_1 dt_1 \right] -$$

$$\left[\int_{T_{overall}}^{\infty} f_{IR}(t_1) T_{overall} \, dt_1 + \int_{0}^{T_{overall}} f_{IR}(t_1) t_1 dt_1 \right] \quad (4.30)$$

Substituting the value of $f_{IR}(t_1)$ from (4.23) in (4.30), we get

$$Loss_{th\_nor} = \frac{1}{\beta}\left(e^{-\beta T_{overalll}} - e^{-\beta T_D'}\right) > 0 \;\forall\; T_{overall} < T_D' \quad (4.31)$$

Thus, it is clearly observed that the proposed approach results in a loss in communication duration and this is the price paid for maintaining the QoS for VoIP services.

Next, the benefits of the proposed algorithm are realized analytically.

(iii) <u>Reduction in Interference</u>

The first objective of reduction in interference with PU (as stated in Section 4.5.1) is successfully met after implementation of the proposed algorithm. Let $Gain_{if\_md}$ be the gain in reduction of interference after implementing the algorithm under the condition of miss-detection. The corresponding expression is derived as follows.

$$Gain_{th\_md} = \left[\int_{T_D'}^{\infty} f_{BR}(t_1) T_D' dt_1 + \int_{0}^{T_D'} f_{BR}(t_1) t_1 dt_1 \right] -$$

$$\left[\int_{T_{overall}}^{\infty} f_{BR}(t_1) T_{overall} \, dt_1 + \int_{0}^{T_{overall}} f_{BR}(t_1) t_1 dt_1 \right] \quad (4.32)$$

Substituting the value of $f_{BR}(t_1)$ from (4.22) in (4.32), we have

$$Gain_{th\_md} = \frac{1}{\alpha}\left(e^{-\alpha T_{overall}} - e^{-\alpha T_D'}\right) > 0 \;\forall\; T_{overall} < T_D' \quad (4.33)$$

Thus, non-effective communication duration and hence, the interference duration are reduced in the miss-detection scenario as the proposed algorithm is implemented, thereby reducing the packet loss and maintaining the QoS for VoIP transmissions. Thus the first objective as stated in Section 4.5.1 is fulfilled.

(iv) ***Reduction in Delay***

Delay reduction is one of the significant criteria for effective VoIP implementation in CRN. One primary factor is delay due to spectrum handoff that occurs with PU arrival in SU transmission time slot and this delay increases with increase in number of switches from current to next idle channel. Let $C_{switch\_perfect}$, C_{switch_base} and C_{switch_algo} refer to the scenarios describing an ideal T_D' that is free from any PU activity, T_D' with PU activity and $T_{overall}$ respectively. Let the VoIP call duration be denoted by T_{call} . Therefore, the expressions for the three scenarios are derived as follows.

$$C_{switch\_perfect} = \frac{T_{call}}{T_D'} \;\forall\; P_{PU} = 0 \tag{4.34}$$

$$C_{switch\_base} = \frac{T_{call}}{T_D'(1 - P_{PU})} \;\forall 0 < P_{PU} < 1 \tag{4.35}$$

$$C_{switch\_algo} = \frac{T_{call}}{T_D'(1 - P_{PU})}\left(1 - P_{t_{propfinal} > t_{thresh}}\right) \tag{4.36}$$

where $P_{t_{propfinal} > t_{thresh}}$ = average probability that $T_{propfinal}$ is greater than the threshold time t_{thresh} , t_{thresh} $=n*$ avg. VoIP talk-spurt time as per the algorithm.

It is, hence, observed that

$$C_{switch\_base} > C_{switch\_perfert} ; C_{switch\_algo} < C_{switch\_base}. \tag{4.37}$$

Thus, the proposed algorithm records less number of channel switching instances. For every channel switch, channel contention time and channel waiting time increase, resulting in higher delays. Hence, in the proposed algorithm, this delay is reduced and VoIP QoS is improved. Thus, the second objective as discussed in Section 4.5.1 is fulfilled.

4.5.4 Implementation in OPNET Simulation Model

In order to verify the analytical inferences, the algorithm is implemented in simulation by developing a model for VoIP over CRN in OPNET Modeler 16.0.A [4.21] based on the design aspects as discussed in Chapter 3. All the significant parameters in VoIP and CRN domain are incorporated in the model along with customizable nodes for further analysis. At first, it is observed from Fig. 4.22 that the global spectrum utilization decreases with time after implementing the algorithm and this confirms the outcome as derived analytically in Section 4.5.3 (i).

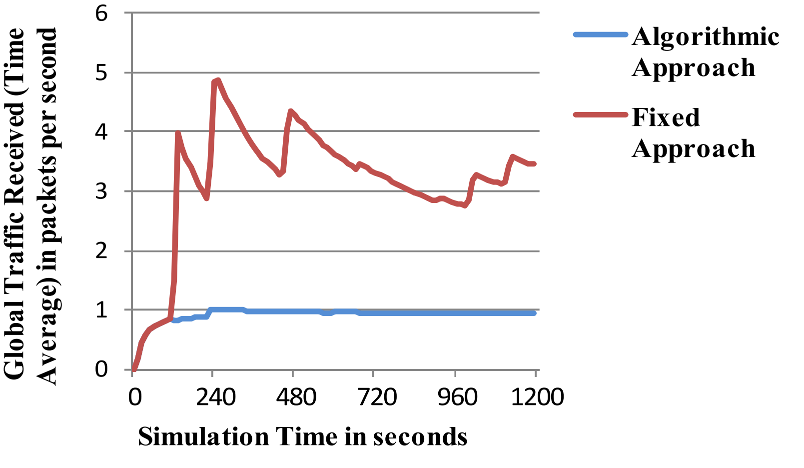


Fig. 4.22 Variation in spectrum utilization in terms of traffic received globally for fixed and algorithmic scenarios during ongoing VoIP call

However, even with lesser throughput, the algorithm achieves lower collision rates and this is witnessed in Fig. 4.23 which is a screenshot of the actual data generated after VoIP transmission in simulation.

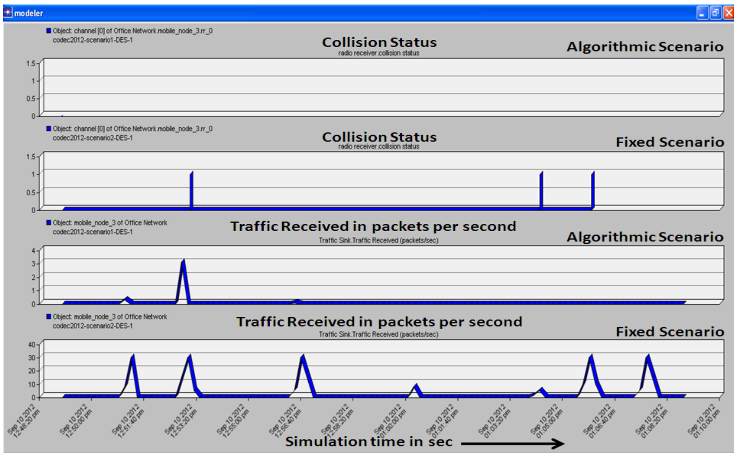


Fig. 4.23 Variation in traffic received along with collision status for fixed and algorithmic scenarios during ongoing VoIP call

Moreover, the mathematical analysis of Section 4.5.3 (iv) is validated in Fig. 4.24 which records decrease in channel switching instances (and hence lesser delay) after application of the algorithm.

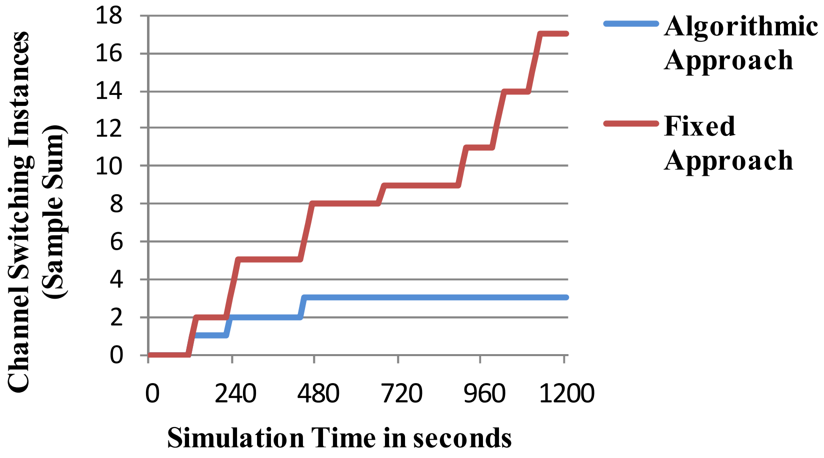


Fig. 4.24 Variation in channel switching instances for fixed and algorithmic scenarios during ongoing VoIP call

Finally, it is concluded from Fig. 4.25 that the developed algorithm adapts itself to increasing PU activity to maintain overall VoIP QoS.

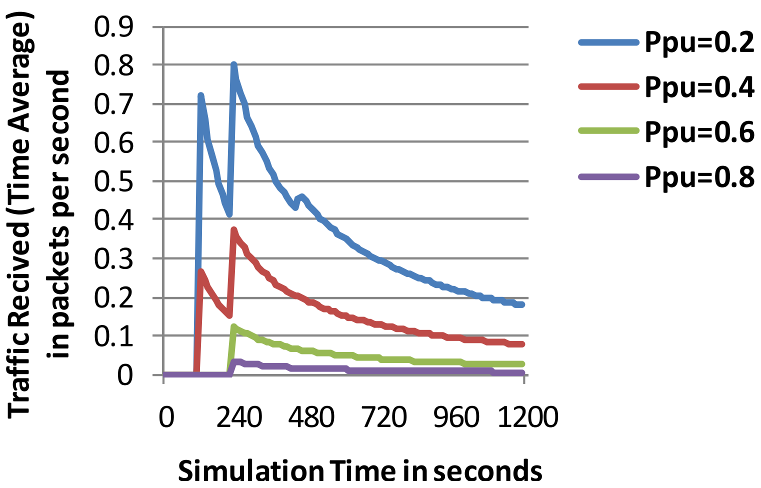


Fig. 4.25 Total Traffic received with increasing PU arrival probability for fixed and algorithmic scenarios

Thus, the simulation outcome in this section not only validates the analytical inferences but also proves the efficiency of the algorithm in satisfying the objectives of Section 4.5.1.

4.6 Design and Application of QoS Metric for evaluating VoIP performance over CRN

Once the different strategies are devised for making the CR timing cycle worthy of supporting VoIP calls, the next step is to evaluate their performance using a derived QoS metric as designed in this section, that will incorporate the features of both VoIP and CRN domain.

Quality is defined as "The totality of features and characteristics of a product or a service that bear on its ability to satisfy stated or implied needs" [4.22]. A distributed system that provides QoS guarantees to general applications must be based on a QoS framework that consists of a QoS specification taxonomy, and a QoS architecture that integrates the different components in the various layers of the system [4.23]. Therefore, proper QoS requirements for the supported applications must be understood before implementing any mechanism.

VoIP has its own QoS parameters that judge the quality of the call by taking into account the network and user conditions as well as environmental factors. In the field of CRN, the overall performance of VoIP is also judged by the amount of opportunistic spectrum utilization. However, many current network architectures address QoS from a provider's point of view and analyze network performance, failing to comprehensively address the quality needs of applications [4.23]. Thus QoS monitoring in VoIP over CRN requires mechanisms to include the overall performance of both CRN and VoIP. The objective is to design a new QoS parameter that takes into consideration the effect of both VoIP and CRN entities.

QoS characteristics can be defined in terms of other characteristics in two ways namely, by specialization of the existing parameters and by derivation and combination of few parameters [4.24]. As per the objective, a derived QoS parameter is defined in this section by taking the combination of both VoIP and CRN parameters to give an indication of the quality of VoIP performance in CRN. As adaptivity is one of the keywords in CRN [4.25], sensing and transmission time parameters are devised as being adaptive to the changing PU traffic in this work (through the algorithm design in Section 4.2). Hence in such an adaptive scenario, the variation in voice quality with channel occupancy is mapped onto a newly designed parameter *cog_cap* which is a measure of the "cognitive capacity" of VoIP users in CRN. It is worth mentioning that as VoIP QoS parameters are unique for a particular VoIP call, *cog_cap* is unique for VoIP session with respect to a particular channel and measures the quality of VoIP transmission over that particular channel.

The metric is defined using a design algorithm. Then this metric is applied to measure the efficiency of the proposed algorithms in this chapter.

4.6.1 Proposed QoS Design Algorithm

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

4.6.2 Discussion of the Algorithm

R-Factor is calculated according to the formula corresponding to the E-model and is given by (4.38) [4.26].

$$R = R_0 - Id - Ie \tag{4.38}$$

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

Channel Occupancy Percentage is calculated by the following equation.

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

(4.39)

Correlation Coefficient has to be determined as per the nature of the curve with respect to S and R. In case of linear ones, Pierson Correlation Coefficient for two variables X and Y can be used which is calculated according to the following equation [4.27] for sample size of N.

$$r = \frac{\sum(X \times Y) - \frac{\sum X \times \sum Y}{N}}{\sqrt{\left(\sum(X)^2 - \frac{\left(\sum X\right)^2}{N}\right)} \times \sqrt{\left(\sum(Y)^2 - \frac{\left(\sum Y\right)^2}{N}\right)}} \tag{4.40}$$

Strength determines the efficiency of r. The higher the value, the more the parameters are correlated to each other. Significance, on the other hand, indicates the validity of the obtained results and strongly depends on the sample size. If it is 0.05, the results are deemed insignificant and hence the sample size must be increased. Hence the value of r must be greater than the critical value of r corresponding to the significance of 0.05. Strength of r is determined by (4.41).

$$s = |r|^2 \tag{4.41}$$

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

4.6.3 Application of the Designed QoS Metric

The designed QoS parameter, *cog_cap* is hence applied to the proposed CR timing cycle algorithm of Section 4.2 in order to measure the cognitive capability of the VoIP user that is implementing the adaptive policy. Initially, the R-Factor and the channel utilization percentage at various instants of an ongoing call are measured and after proper scaling, their values are mapped onto a graph. As the trend is almost linear as observed from Fig. 4.26, Pearson's correlation coefficient is applied as per the proposed algorithm.

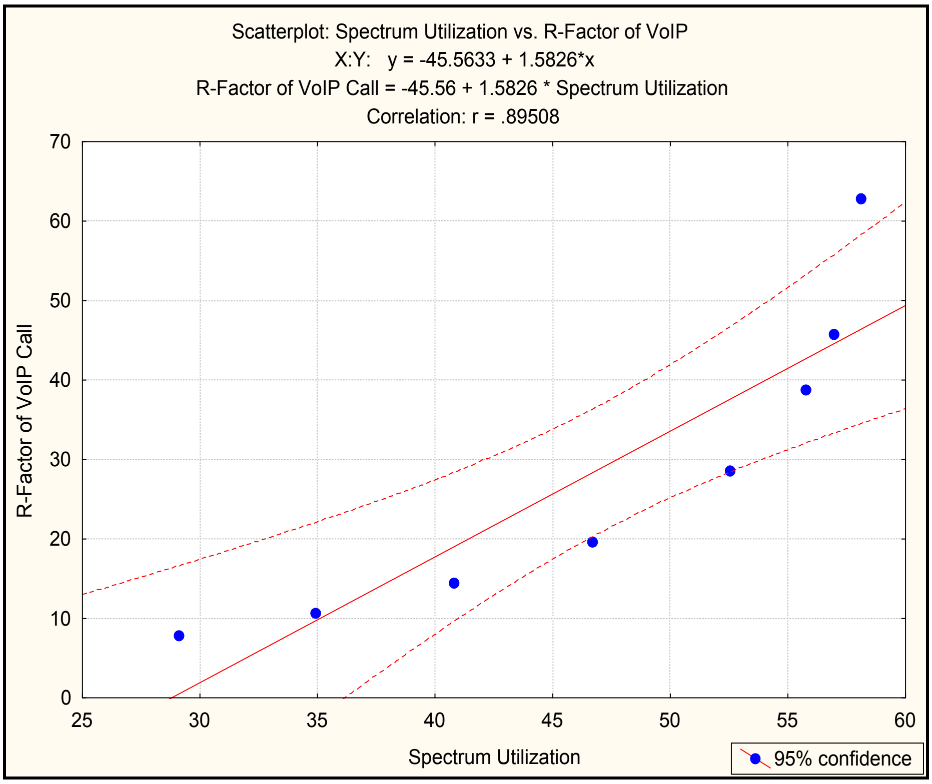


Fig. 4.26 Calculation of *cog_cap* for the proposed adaptive algorithm

It is observed from the figure that the value of *cog_cap* is 0.89 having strength 0.79. This indicates that as the channel utilization increases, there is 79 percent increase in VoIP call quality. The critical value of *r* corresponding to statistical significance of 0.05 for a sample size of 8 is 0.707 which is less than the obtained value of 0.89. Hence, the obtained result is statistically significant.

Thus, the derived QoS metric proves highly useful for jointly evaluating the performance of VoIP and CR parameters.

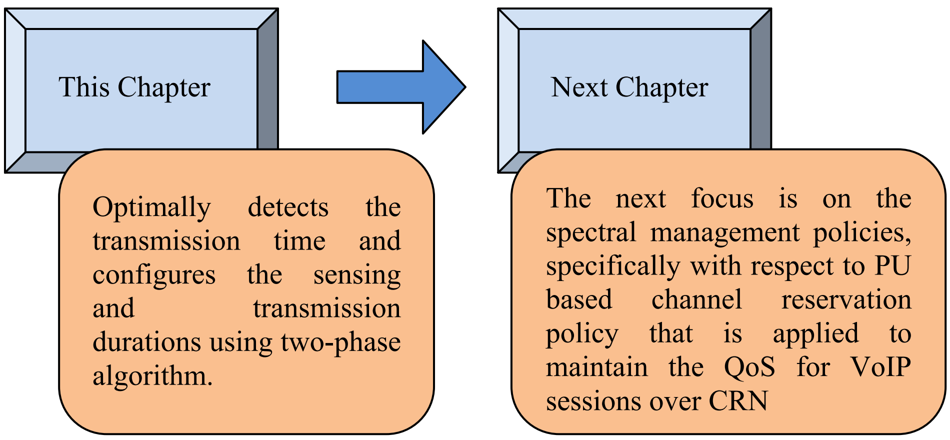
4.7 Summary

Considering the inadequacy of the basic CR timing cycle in providing QoS protection to the VoIP traffic as generated by the SUs, this chapter addresses the problem of sensing-throughput tradeoff as prevalent in CRN and accordingly formulates a novel CR cycle modification algorithm in two steps. In the first part, momentary sensing slots are incorporated in the transmission interval that help in providing maximum time for transmission by VoIP SUs without interfering with the PUs. Thereafter in the second part of the algorithm, the CR timing cycle is adaptively varied using feedback parameters depending on PU activity pattern. This methodology increases channel utilization and call quality for VoIP SUs during low PU activity and also protects PU transmissions when they are highly active. Extensive analysis of the proposed enhancements in the real life-like simulation models of OPNET Modeler bears a clear testimony to the fact that even with zero tolerance for signal loss with respect to the PUs, the quality of VoIP communication is retained for the SUs. Both the end-to-end delay and jitter remain within 60 ms and 50 ms respectively under low PU activity and also below 100 ms and 80 ms respectively in case of high PU activity. Also, the throughput increases for SUs in both these cases that indicate better channel utilization. Overall, all the QoS metrics record a drastic improvement after modifying the basic CR timing cycle with the proposed algorithm. Thereafter, mathematical studies of the algorithm reflect the importance of optimal selection of related parameters (sensing, transmission intervals, feedback parameters, detection probability) towards attainment of the overall objective. In this regard, the essential conditions are derived from the mathematical framework, which will guide towards configuring the related parameters.

Additionally, this work emphasizes on the aspect of detecting optimal transmission duration before initiating VoIP sessions in the designated channel. In this regard, a novel algorithm is designed that carefully eliminates the effect of imperfect channel sensing and carefully chooses optimal transmission time for successful VoIP communication. Simulation outcome in the OPNET models record improved call quality although at the cost of low channel utilization. The same inference is also validated using an analytical framework that establishes the credibility of the proposed technique.

Finally, in the last module of this chapter, a derived QoS parameter denoted by *cog_cap* is designed that measures the cognitive capacity of the VoIP calls. Application of *cog_cap* to the proposed CR timing cycle algorithm records 79% increase in R-Factor as the channel utilization improves through increased transmission opportunities for VoIP based SUs. This confirms the overall applicability of the proposed design methodologies in this chapter.

The outcome of this study has been published in ***International Journal of Computer Information Systems and Industrial Management Applications, Machine Intelligence Research (MIR) Labs'14*** (**Elsevier SCOPUS Indexed**) and also in the ***Proceedings of IEEE WICT'11 (Mumbai) and IEEE CODEC'12 (Kolkata)***.



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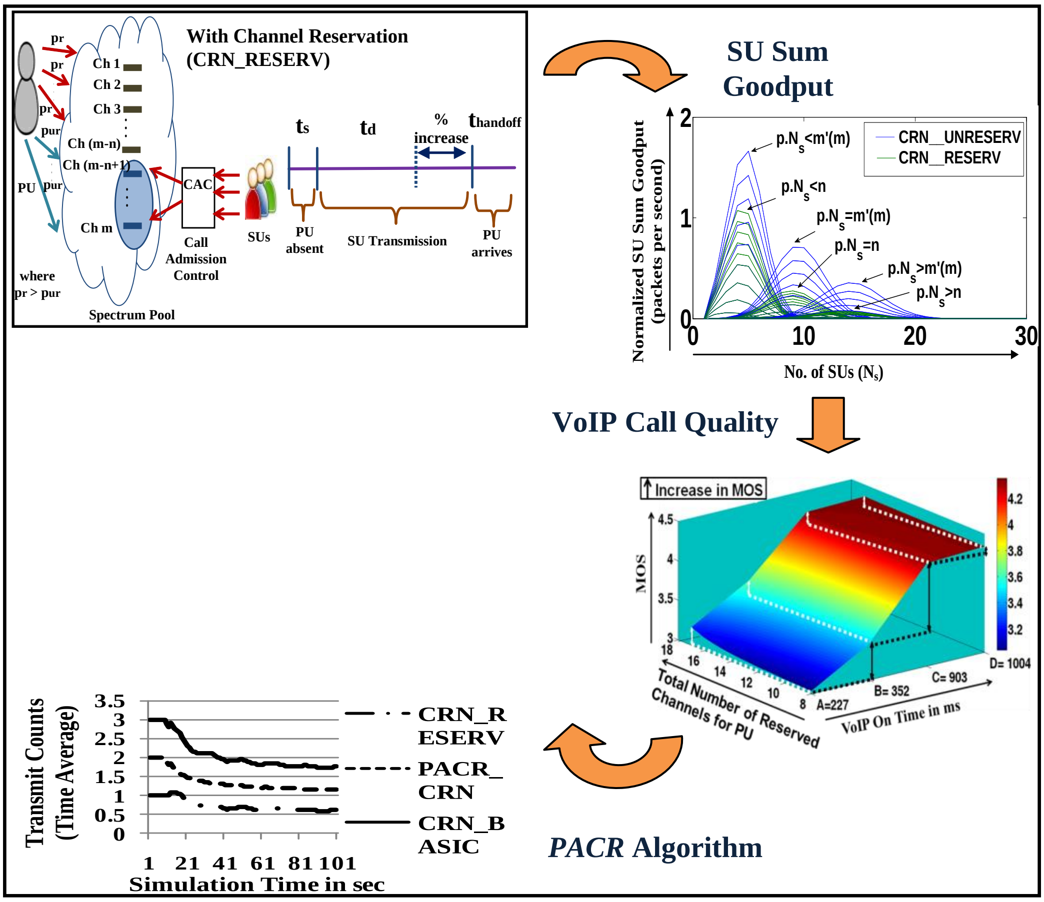
Chapter 5.

CHANNEL RESERVATION SCHEME FOR VOIP USERS IN CRN

Chapter Highlights



CHAPTER 5: Channel Reservation Scheme for VoIP Users in CRN

"We will have more Internet, larger number of users, more mobile access, more speed, more things online and more appliances we can control over the Internet"

- Vinton Cerf, Father of TCP-IP

Outline of the Chapter

After suitably configuring the timing parameters of each CR cycle for VoIP transmission in the previous chapter, the next focus is on applying spectrum sharing and management policies over VoIP applications. Specifically, this chapter deals with channel reservation policies in order to ensure maximum throughput for these SUs, while minimizing their interference with PUs. While few research studies have designed Markov Models and applied heuristic approaches with respect to reserving channels for either SUs or PUs, they did not address the practical constraints involved in implementing such a system. Lack of a proper mathematical model further limits extended work in this domain. Additionally, the efficiency of this mechanism has not been studied from the perspective of the applications that are highly sensitive to delay and loss, such as VoIP. ***All these issues lay the foundation towards a***

complete study of channel reservation policy in this Chapter with its practical considerations and specific advantages towards real-time VoIP applications.

Initially, the system model is defined along with an architectural framework to address the practical constraints involved in implementing the channel reservation strategy. A novel design approach is then adopted to develop the mathematical models, followed by an evaluation of the network parameters to obtain a tradeoff between the decrease in interference and low spectrum utilization when some channels are reserved in the CRN. The optimal number of channels to be reserved is also determined. In addition, as a potential application of the channel reservation policy, an analytical framework is designed for the VoIP applications in the CRN, that highlights significant improvement in the VoIP call quality. As reserving fixed number of channels for PUs lead to under-utilization of idle channels in times of low PU activity, this chapter further introduces a novel *Priority based Adaptive Channel Reservation* (*PACR*) algorithm that adaptively reserves channels based on PU activity and performs priority based channel allocation to both real-time and non real-time users. Mathematical models are designed for *PACR* based CRN, that depict increase in system capacity and system heterogeneity with respect to hosting both VoIP and data applications. Finally, an extensive analysis in real-life like simulation models validates the inferences drawn from the mathematical models, recording over 50% improvement in the interference-free transmission of the SUs with the channel reservation scheme in the CRN. *PACR* ensures another 100% improvement in SU throughput as compared to static channel reservation policy.

5.1 Introduction

Channel reservation policy addresses the issues of interference on the sudden PU arrival and subsequent handoff by the SUs. Unlike the principle of providing the reserved channels to the mobile users in the traditional wireless networks [5.1], the channel reservation strategies in the CRN can be categorized with respect to the SUs and the PUs. The SU based channel reservation schemes focus on reserving the idle channels for the SUs in both single and multi-cooperative CRNs with the aim of maintaining an optimal trade-off between the blocking and dropping probabilities and have been widely studied in [5.2-5.14]. The PU based channel reservation policy, on the other hand, has witnessed limited research [5.15-5.18] and forms the basis of the work in this chapter. In this scheme, a number of idle channels in the CRN are reserved for the PUs. On arrival, the PUs utilize the reserved channels. Once the reserved channels are occupied, these PUs access rest of the idle channels in the CRN. Hence, the reserved channels are never allocated to the SUs. The SUs can access only the unreserved idle channels. This ensures interference-free transmission by these SUs for a longer duration of time and reduces the possibility of spectrum handoff, as the probability of arrival of a PU on a channel already occupied by an SU decreases.

In this regard, a detailed literature survey is conducted in the following section in order to observe the advantages that such policies can provide to real-time communication

5.1.1 Literature Survey

QoS analysis and management with suitable spectrum sharing policies in CRN is an area of great concern [5.19]. Accordingly, this section provides an in-depth study of the existing works in the domain of channel reservation policy as applied to CRN, and highlights the pros and cons of both PU and SU based channel reservation. Consequently, the novelty of this work is highlighted.

(i) <u>Literature Survey: SU based Channel Reservation Policy</u>

These works have focused on reserving some channels for the SUs [5.2-5.14]. In this system, the SUs perform transmission in the idle channels when the PUs are absent. Any PU arrival in the current channel forces the SU to perform spectrum handoff and shift to the reserved channels. Performance improvement is recorded with respect to the call dropping and blocking probabilities for the SUs. The QoS parameters are also studied with respect to spectrum handoff by the SUs to the reserved channels [5.2, 5.8]. This is followed by complexity analysis of reserving the channels in a CRN [5.13]. As a potential application, the SU based channel reservation scheme has been used to prioritize the service requirements of the existing SUs over new SUs in [5.14]. However, reserving channels for the SUs has two major drawbacks.

Firstly, it is difficult to implement channel reservation in an overlay based CRN. This is because in absence of any cooperation between the PUs and SUs, a PU can access every licensed channel with an equal probability. As a result, there is a high probability that a SU may find a reserved channel occupied by a PU during spectrum handoff, leading to severe degradation in performance. Secondly, SU based channel reservation does not ensure any reduction in spectrum handoff for the SUs and is, therefore, not suitable for those applications that are highly sensitive to frequent interruptions in transmission, such as VoIP applications.

(ii) ***Literature Survey: PU based Channel Reservation Policy***

The PU based channel reservation scheme [5.15-5.18] addresses the problems incurred in reserving the channels for the SUs. In this system, a set of channels is reserved for the PUs to access and transmit. The SUs can only occupy the unreserved idle channels. The PUs gain access to the unreserved channels after occupying all the reserved channels. The advantage is that it significantly delays the arrival of a PU in the channel occupied by a SU, thereby reducing the probability of SU spectrum handoff and related complexities. Therefore, this policy is highly suited to the scenarios where spectrum handoff has to be minimized. In this respect, a tradeoff is achieved in [5.15] between low dropping probability and high blocking probability for the SUs, that is extended in [5.16] with the incorporation of spectrum handoff. Moreover, the guard channel policy, where a number of channels is dedicated to the primary network calls, has been investigated in [5.17] to reduce the blocking probability for the PUs and SUs. This work has been further extended in [5.18] where both PUs and SUs operate in a similar service area and utilize the same single cell.

It is clearly observed from the literature survey that the PU based channel reservation scheme has received lesser attention than the SU based channel reservation policy. Also, these works suffer from several drawbacks that serve as the primary motivation behind the work in this chapter. Table 5.1 summarizes the limitations of these studies and subsequently highlights the contributions of this chapter.

Table 5.1 Significance of the Proposed Work with respect to the Previous Works

Research Aspects	Previous Works on PU based Channel Reservation in the CRN	Work in this Chapter	Significant Contributions of this Chapter
1. System Model	The Markov Model maps the interaction between a PU and a SU in [5.15, 5.16]. The Heuristic Approach models the reservation of guard channels for the PUs in [5.17, 5.18]. **Limitation:** Difficult to incorporate varying traffic distributions for both PU and SU, omits several critical aspects of the system design.	The Mathematical Model in Section 5.4 includes both PU and SU ***traffic characteristics along with imperfect channel sensing factors.*** The Architectural Model in Section 5.3 provides the ***solutions for practical deployment.*** The Simulation Model in Section 5.7 provides the ***validation of analytical output*** and also provides real life-like implementation of PU based channel reservation. **Advantage:** A more ***realistic modeling*** of the PU based channel reservation.	**A Comprehensive Design of a Robust System Model based on Analytical Framework, Practical Deployment and Simulation Platform**.
2. System Parameters	Arrival Rate and Mean Service Time for PU and SU, Total no. of PUs and SUs, Channel state conditions, dropping probabilities [5.15, 5.16], blocking probabilities [5.15, 5.16], Spectrum Handoff [5.16], No. of reserved channels [5.17, 5.18]. **Limitation:** Lack of analysis of the critical system parameters, difficult to derive the optimal tradeoff among conflicting	SU and PU Arrival Probability, No. of SUs and PUs, Imperfect channel sensing parameters, No. of reserved channels, Dropping, blocking, handoff and transmission probabilities in Section 5.4, Channel Throughput, Spectrum Handoff delay, Time of interference, Application-level parameters in Section 5.7. **Advantage:** All the system parameters are mathematically derived, ***optimal tradeoff is achieved*** among the conflicting system parameters after rigorous stochastic analysis, the optimal number of channels	**A Complete Analysis of all the System Parameters along with the derivation of an optimal tradeoff among the conflicting parameters.**

	system parameters.	to be reserved is a key factor for successful implementation of channel reservation and is derived and verified. Also, priority based adaptive reservation is performed.	
3. Practical Aspects	Implementation aspects are not considered in any of these works. **Limitation:** Suffers from the practical deployment issues.	The Architectural Framework is designed for the overall CRN in Section 5.3 that includes both the system information and the channel state information. **Advantage:** Implementation issues reflect proper coordination among several domains of CRN and provide ***practical solution for channel reservation.***	**Design of a Generic Model for all traffic and further extending it to a real-time VoIP Application Model.**
5. Performance Evaluation	Call dropping and blocking probability with arrival rates [5.15, 5.16] and no. of reserved channels [5.17, 5.18], Allowable traffic in CRN [5.15], Handoff Issues [5.16]. **Limitation:** Requires thorough analysis by recording the variation with different parameters, Lack of an application-level parameter analysis.	Both the system-level parameters (SU Sum Goodput, SU dropping instances, Interference, dropping, blocking, handoff and transmission probabilities) and the application-level parameters (throughput, spectrum handoff delay, packet loss, link utilization, Mean Opinion Score) along with RT and NRT User characteristics are evaluated in Section 5.4 and Section 5.7. **Advantage:** A complete study of parameters from the ***system and application perspective***, more rigorous evaluation of the system performance with respect to the previous works	**A Complete Analysis of CRN performance with respect to both Analytical and Simulation results and detailing every aspect of the system level performance and individual VoIP applications in conjunction with the non real-time users.**

5.1.2 Significant Contributions

Accordingly, the objective of this chapter is to perform a complete analysis of the PU based channel reservation policy in the CRN with the design of suitable generic mathematical models after taking into account the practical constraints and evaluation of the system performance with respect to the real-time applications.

The major contributions of this chapter are summarized as follows. Also, the novelty of this work is broadly explained in Table 5.1.

1. An extensive survey of the ongoing research in Section 5.1 reveals lack of a complete study of channel reservation in the CRN. Accordingly, the objective of this work is clearly highlighted in Section 5.1.
2. The system model for the CRN with channel reservation is described in Section 5.2 along with the design of an architectural framework in Section 5.3 to discuss the constraints involved in implementing such strategies in the practical systems.
3. The analytical models are designed for the basic CRN and the CRN with channel reservation in Section 5.4 followed by the derivation of an optimal tradeoff among the conflicting system parameters and comparative performance evaluation.
4. The channel reservation strategy is implemented in a VoIP based CRN in Section 5.5. The performance parameters related to the call quality are derived after the formulation of the VoIP traffic and design of the mathematical models.
5. In order to address the basic problems in static channel reservation policy (loss in utilization and lack of QoS protection), a novel *Priority based Adaptive Channel Reservation* (*PACR*) algorithm is designed in Section 5.6 and analyzed extensively in analytical models.
6. Finally, detailed performance evaluation with respect to the channel reservation policy is carried out in the analytical and simulation models in Section 5.7 at the user level (with respect to the call quality) and the system level (in terms of overall gain in the system performance).

Finally, the chapter is concluded in Section 5.8.

To the best of our knowledge, no such work covering all the details of PU based channel reservation policy in a CRN has been reported so far.

5.2 System Model Design with Channel Reservation Policy

A basic CRN model comprises of both PUs and SUs. Every PU is allotted a licensed spectrum band where it performs transmission for a certain period of time. However, a PU may not transmit continuously. A SU senses the band for PU presence in the sensing time interval (denoted by t_s). If the PU is sensed to be absent, the SU starts its transmission in that band for the SU transmission time interval (denoted by t_d) [5.20]. In the absence of any channel reservation policy, a PU can access any channel on arrival and start its transmission, thereby leading to interference with any ongoing SU transmission in the same channel. The SU must suspend all the activities and perform spectrum handoff to move to an available idle channel. When all the channels are occupied, any communication by that SU has to be dropped. Therefore, CRN without any channel reservation policy (denoted by CRN_UNRSERV) experiences severe degradation in the quality of transmission for both the PUs and SUs. CRN_UNRESERV is depicted pictorially in Fig. 5.1.

As the PU based channel reservation policy is applied to the CRN, a suitable number of idle channels is initially reserved for the incoming PUs. The SUs enter the CRN following a certain Call Admission Control policy and can access only the unreserved idle channels. This decreases the overall number of SUs to be supported by that CRN. As long as the CRN has available reserved channels, the PUs cannot occupy the unreserved channels. Therefore, the probability of accessing an unreserved channel by a PU decreases in this scenario. Thus, the SUs continue transmission in the unreserved channels for a longer duration of time, resulting in a significant reduction in the spectrum handoff and dropping probabilities. Moreover, the interference among these users decreases due to prioritized PU access in the reserved channels. Hence, a CRN with the channel reservation policy (denoted by CRN_RESERV) records reduced dropping and handoff probabilities and lower levels of interference with decrease in the total number of SUs in the network. CRN_RESERV is depicted pictorially in Fig. 5.1.

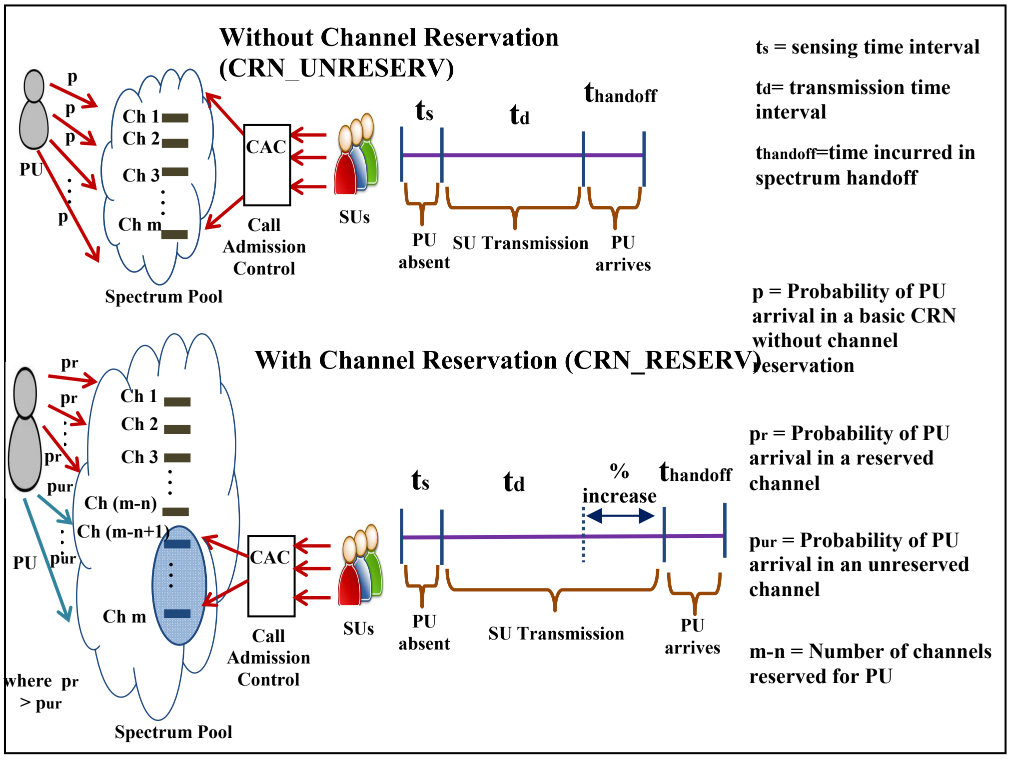


Fig. 5.1 System Model for CRN_UNRESERV and CRN_RESERV

5.3 Architectural Framework

The practical aspects of implementing CRN_RESERV require several design considerations that must be carefully addressed with respect to every node in the CRN. Firstly, the CRN must have a spectrum broker (or a spectrum controller node) [5.20] that implements the channel reservation scheme for the PUs and thus, serves as the most significant element in CRN_RESERV. It determines the optimal number of channels to be reserved for the PUs. Accordingly, it informs the PU and SU controller nodes regarding the valid channels that can be used by the corresponding PUs and SUs and also maintains proper coordination between the controller nodes. The controller nodes grant access of the available marked idle channels to the corresponding PUs and SUs, respectively. Once the PUs have occupied all the reserved channels, they are eligible to access other available idle channels with equal probability. The SUs in that case must perform spectrum handoff and CRN_RESERV resembles CRN_UNRESERV in such scenarios.

Thus, the overall allocation of channels in CRN_RESERV can be described as a 4-step procedure where,

1. The spectrum broker executes channel reservation for the PUs and informs both the PU and SU controller nodes.
2. The PUs and SUs implement the MAC protocols to be eligible to access the idle channels.
3. The controller nodes for both the PUs and SUs perform channel assignment to allocate the most efficient channel to the available users.
4. The SUs perform spectrum handoff on PU arrival, once the reserved channels are fully occupied by the PUs.

The architectural framework for this process is shown as a schematic diagram in Fig. 5.2. This figure validates the contribution of this work in terms of providing a practical solution for deploying CRN_RESERV as mentioned in Table 5.1. It can be inferred that the successful implementation of PU based channel reservation requires significant contributions in several domains of the CRN. This chapter only focuses on designing the channel reservation policy that will provide the necessary platform for further research on MAC protocols, channel assignment schemes and handoff strategies.

5.4 Design and Analysis of Analytical Framework for CRN_RESERV

This section develops the mathematical models of CRN for two scenarios where i) no channels are reserved, and ii) some idle channels are reserved for the PUs. Initially, a wireless channel is established mathematically followed by the design of analytical models for CRN_UNRESERV and CRN_RESERV, respectively.

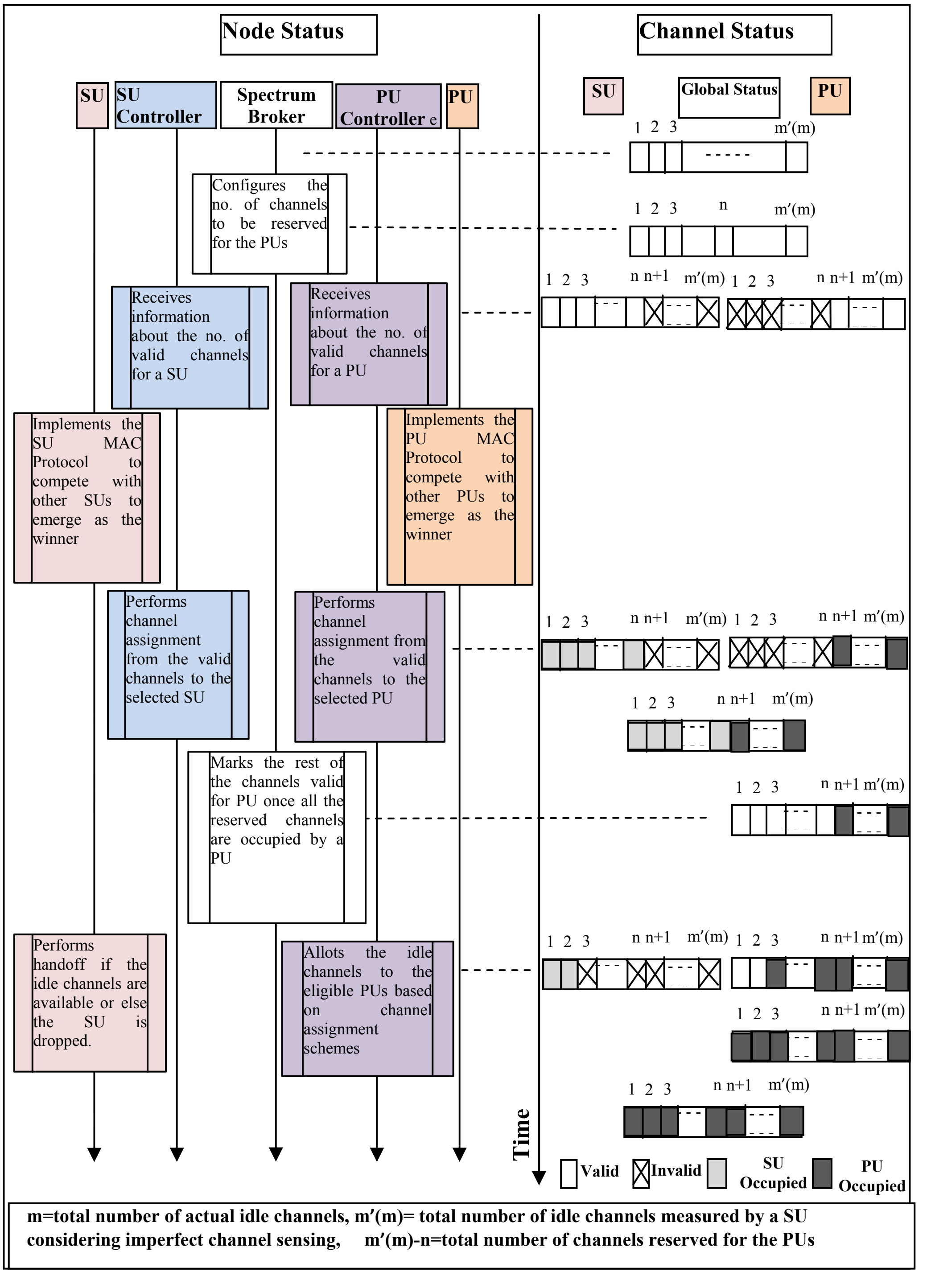


Fig. 5.2 Schematic Diagram of an Architectural Framework required for implementing PU based Channel Reservation Policy in CRN

5.4.1 Design of a Wireless Channel

A wireless channel is designed as a two-state Markov process as depicted in Fig. 5.3. An *Occupied State* suggests that the current channel is occupied by a PU. An *Unoccupied State* refers to an idle channel that is not occupied by a PU and hence, can be accessed by a SU. Let the system comprises of M wireless channels. Therefore, the transition probability ($P_{m,n}$) that there are m unoccupied channels in the current frame and n unoccupied channels in the next frame is given by [5.21].

$$P_{m,n} = \sum_{x'=\max(0,m-n)}^{\min(m,M-n)} \binom{m}{x'} p_{01}^{x'} p_{00}^{m-x'} \binom{M-m}{y'} p_{10}^{y'} p_{11}^{M-m-y'} \tag{5.1}$$

where $y'=n-m+x'$ and x' and y' denote the number of channels whose status have changed from *Unoccupied* to *Occupied* and *Occupied* to *Unoccupied*, respectively.

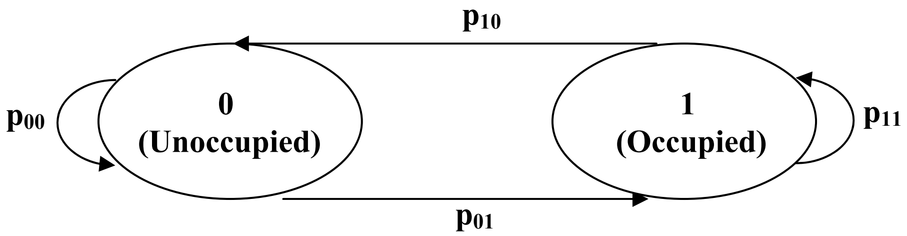


Fig. 5.3 Markov Model for a single wireless channel

However, imperfect spectrum sensing introduces the problems of false alarm and miss-detection [5.21]. Therefore, the number of measured unoccupied channels is a function of the total number of actual unoccupied channels, that is denoted by $m'(m)$ and is given by,

$$m'(m) = m - m \times p_f + (M-m) \times (1-p_d) \tag{5.2}$$

where, p_f = probability of false alarm, p_d = probability of detection and $1-p_d$ = probability of miss detection.

5.4.2 Modeling the CRN with no Channel Reservation (CRN_UNRESERV)

The model for CRN_UNRESERV is established with respect to the SU Sum Goodput that is defined as the total amount of the SU data successfully delivered per unit time. Let N_s be the total number of SUs, each transmitting at a rate R with probability p and let N_p be the total number of PUs that transmit with the same probability p. Therefore, total number of active PUs is given by, $\sum_{i=1}^{N_p}\binom{N_p}{i}p^{N_p-i}(1-p)^i$. Thus, there are '$i$' number of idle channels to be allotted to 'j' number of SUs that are again active with the probability $p^j(1-p)^{N_s-j}$. Accordingly, the SU Sum Goodput ($C_S^{sum}(i)$) for the CRN with no channel reservation is given by [5.22].

$$C_s^{sum}(i)=\sum_{i=1}^{N_p}\binom{N_p}{i}p^{N_p-i}(1-p)^i\sum_{j=1}^{N_s}\binom{N_s}{j}p^j(1-p)^{N_s-j}Rj\left(1-\frac{1}{i}\right)^{j-1} \quad (5.3)$$

As every PU is allotted a licensed channel, the total number of channels in the system is equal to the total number of PUs and is given by N_p. It is observed from (5.3) that at any particular instant, N_p-i number of PUs are available for transmission with probability p. Hence, the number of idle channels is given by i. ((N_p-(N_p-i)) = i). At that instant, j number of SUs is active in the CRN with probability p. The probability of j SUs accessing i number of idle channels is illustrated in (5.3).

The basic considerations in (5.3) are eliminated in this chapter to present a generalized scenario that is discussed as follows.

1. Instead of having equal PU and SU arrival probabilities as in (5.3), different arrival probabilities for PUs and SUs as denoted by p_r and p_s, respectively, are considered in this chapter to model different traffic distributions for these users.
2. The idle channels (equal to i) are uniform in every scenario in (5.3). On the contrary, this chapter models the total number of idle channels after an appropriate design of the wireless channel, that is denoted by $m'(m)$.
3. Finally, the data transmission rate corresponding to each SU is considered identical as given by R in (5.3). This work considers different transmission rates for the SUs. R_q denotes the transmission rate corresponding to the qth channel where q belongs to $m'(m)$.

Hence, the SU Sum Goodput is modified and expressed as in (5.4). It is observed from (5.4) that the PU arrival probability p_r remains constant for every scenario that describes channel allocation to the SU. The throughput for each

SU per unit time in every such scenario is denoted by C_q. At any instant, N_p-m number of PUs occupies the channels in the CRN with probability p_r. Therefore, the total number of actual idle channels left for the SUs to access is given by m. From the SU perspective, considering imperfect channel sensing, $m'(m)$ number of channels is idle (as given by (5.2)) and can be accessed by j number of SUs that are active in the CRN with probability p_s.

$$C_s^{sum}(m) = \sum_{m=1}^{N_p} \binom{N_p}{m} p_r^{N_p - m} (1 - p_r)^m \sum_{j=1}^{N_s} \binom{N_s}{j} p_s^{\,j} (1 - p_s)^{N_s - j} C_q$$

$$where\ \ C_q = \sum_{q=1}^{m'(m)} \left\{ R_q \binom{m'(m)}{1} \left(\frac{1}{m'(m)} \right) \left(1 - \frac{1}{m'(m)} \right)^{j-1} \right\} \tag{5.4}$$

However, a PU can arrive at any moment with varying p_r. In order to capture the dynamic PU behavior, an event driven approach is followed in this work that incorporates both *initial* and *current* PU activity. The initial PU arrival probability is denoted by p_r. The current status of the PU arrival before the channels are allocated to the SUs is marked as an event. Every such event is expressed in terms of a binary variable defined by,

$$\begin{aligned} H_i &= 1 \quad \forall i\ no.\ of\ \ PUs\ \ present\ \ in\ the\ system \\ &= 0 \quad otherwise \end{aligned} \tag{5.5}$$

The expression for C_q is hence derived from (5.4) as follows.

$$C_q = (1 - H_1) \sum_{q=1}^{m'(m)} \left\{ R_q \binom{m'(m)}{1} \left(\frac{1}{m'(m)} \right) \left(1 - \frac{1}{m'(m)} \right)^{j-1} \right\} \ \forall\ no\ PU\ \ present$$

$$C_q = H_1 \sum_{q=1}^{m'(m)-1} \left\{ R_q \binom{m'(m)-1}{1} \left(\frac{1}{m'(m)-1} \right) \left(1 - \frac{1}{m'(m)-1} \right)^{j-1} \right\} \ \forall\, 1\ PU\ \ present$$

$$\vdots$$

$$C_q = H_k \sum_{q=1}^{m'(m)-k} \left\{ R_q \binom{m'(m)-k}{1} \left(\frac{1}{m'(m)-k} \right) \left(1 - \frac{1}{m'(m)-k} \right)^{j-1} \right\} \forall\ k\ \ PU\ \ present$$

(5.6)

Therefore, the SU Sum Goodput for CRN_UNRESERV is derived after including all the possible events and is expressed as,

No PU Presence: All channels available

$$C^{sum}_{UNRESERV}(m)=\sum_{m=1}^{N_p}\binom{N_p}{m}p_r^{\,N_p-m}(1-p_r)^m\sum_{j=1}^{N_s}\binom{N_s}{j}p_s^{\,j}(1-p_s)^{N_s-j}\left[(1-H_1)\sum_{q=1}^{m'(m)}\left\{R_q\binom{m'(m)}{1}\left(\frac{1}{m'(m)}\right)\left(1-\frac{1}{m'(m)}\right)^{j-1}\right\}\right.$$
$$\left.+\sum_{i=1}^{m'(m)-2}\left\{H_i\sum_{q=1}^{m'(m)-i}\left\{R_q\binom{m'(m)-i}{1}\left(\frac{1}{m'(m)-i}\right)\left(1-\frac{1}{m'(m)-i}\right)^{j-1}\right\}\right\}+H_{m'(m)-1}R_{m'(m)}\right]$$

As PUs arrive, idle channels decrease

(5.7)

5.4.3 Modeling the CRN with Channel Reservation (CRN_RESERV)

Following the principle of channel reservation for the PUs, CRN_RESERV is modeled based on the consideration that *m′(m)-n* number of channels is reserved for the PUs. Thus, the SUs can occupy at the most *n* number of channels out of *m′(m)* idle channels. The expression of C_q with channel reservation for the PUs for different values of H_i is given as,

$$C_q=(1-H_1)\sum_{q=1}^{n}\left\{R_q\binom{n}{1}\left(\frac{1}{n}\right)\left(1-\frac{1}{n}\right)^{j-1}\right\}\quad\forall\ no\ PU\ presence$$

$$C_q=H_1\sum_{q=1}^{n}\left\{R_q\binom{n}{1}\left(\frac{1}{n}\right)\left(1-\frac{1}{n}\right)^{j-1}\right\}\quad\forall\ 1\ PU\ presence$$

$$C_q=H_{m'(m)-n}\sum_{q=1}^{n}\left\{R_q\binom{n}{1}\left(\frac{1}{n}\right)\left(1-\frac{1}{n}\right)^{j-1}\right\}\quad\forall\ m'(m)-n\ PU\ presence$$

$$\vdots$$

$$C_q=H_{m'(m)-n+1}\sum_{q=1}^{n-1}\left\{R_q\binom{n-1}{1}\left(\frac{1}{n-1}\right)\left(1-\frac{1}{n-1}\right)^{j-1}\right\}\quad\forall\ m'(m)-n+1\ PU\ presence$$

$$\vdots$$

$$C_q=H_k\sum_{q=1}^{m'(m)-k}\left\{R_q\binom{m'(m)-k}{1}\left(\frac{1}{m'(m)-k}\right)\left(1-\frac{1}{m'(m)-k}\right)^{j-1}\right\}$$

$$\forall\ k\ PU\ presence\ where\ k>m'(m)-n \qquad (5.8)$$

It is quite evident from (5.8) that as long as PUs restrict themselves to the reserved channels, SUs do not need to perform channel switch operations as the number of idle channels remains fixed at 'n'. Only when PUs start arriving in the unreserved channels, the number of idle available channels start decreasing and SUs need to perform handoff operations. Thus, (5.8) highlights the basic advantage that PU based channel reservation scheme provides to the SUs.

Henceforth, the SU Sum Goodput for CRN_RESERV is derived after incorporating all the events of (5.8) and is expressed in (5.9). It is considered in (5.9) that CRN_RESERV consists of minimum two idle channels so that at least one channel is reserved for the PU.

$$C_{RESERV}^{sum}(m)=\sum_{m=2}^{N_p}\binom{N_p}{m}p_r^{N_p-m}(1-p_r)^m\sum_{j=1}^{N_s}\binom{N_s}{j}p_s^{\,j}(1-p_s)^{N_s-j}\Bigg[\overbrace{(1-H_1)\sum_{q=1}^{n}\left\{R_q\binom{n}{1}\left(\frac{1}{n}\right)\left(1-\frac{1}{n}\right)^{j-1}\right\}}^{\text{No PU Presence: All channels available}}$$
$$+\underbrace{\sum_{i=1}^{m'(m)-n}\left\{H_i\left\{\sum_{q=1}^{n}\left\{R_q\binom{n}{1}\left(\frac{1}{n}\right)\left(1-\frac{1}{n}\right)^{j-1}\right\}\right\}\right\}}_{\text{PU arrives in the reserved channels. SU remains unaffected}}+\underbrace{\sum_{i=m'(m)-n+1}^{m'(m)-2}\left\{H_i\sum_{q=1}^{m'(m)-i}\left\{R_q\binom{m'(m)-i}{1}\left(\frac{1}{m'(m)-i}\right)\left(1-\frac{1}{m'(m)-i}\right)^{j-1}\right\}\right\}}_{\text{PU arrives in the unreserved channels. SU has to perform handoff}}$$
$$+H_{m'(m)-1}R_{m'(m)}\Bigg] \tag{5.9}$$

5.4.4 Evaluation of the Performance Parameters

The developed models of CRN_UNRESERV and CRN_RESERV are analyzed for performance efficiency. The system parameters that are considered for evaluating the performance of channel reservation in CRN include $C_{UNRESERV}^{sum}$, C_{RESERV}^{sum}, N_s, p_s , $m'(m)$ and n.

In order to derive the optimal trade-off among the system parameters in CRN_UNRESERV and CRN_RESERV, the normalized SU Sum Goodput for CRN_UNRESERV and CRN_RESERV, as derived in (5.7) and (5.9), respectively, are plotted in Fig. 5.4 with respect to N_s for different values of $m'(m)$. A rise in the PU activity is modeled by increasing p_r. Hence, $m'(m)$ decreases, thereby symbolizing a network with high traffic load. As observed from Fig. 5.4, CRN_UNRESERV provides higher SU Sum Goodput as compared to CRN_RESERV. This is because with channel reservation, the SUs cannot occupy the idle channels that are reserved for the PUs. Moreover, the SU Sum Goodput decreases with increase in the number of SUs for CRN_ UNRESERV and CRN_RESERV as the probability of accessing an idle channel decreases with rise in SUs. It is obvious from Fig. 5.4 that the trade-off between SU Sum Goodput and the number of supported SUs in CRN_ UNRESERV is optimal under the following condition.

$$N_S \times p_S = m'(m) \tag{5.10}$$

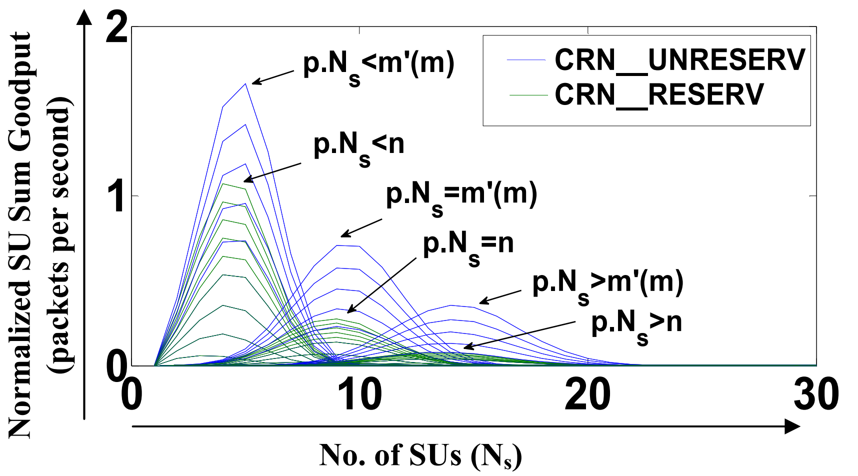


Fig. 5.4 Variation in the Normalized SU Sum Goodput with the total number of SUs in CRN_ UNRESERV and CRN_RESERV for different number of idle channels

It must be noted that (5.10) can be derived from the mathematical model in [5.22], thereby establishing the credibility of the designed analytical model in this chapter. The derivation is provided below.

Validation of the condition $N_s*p_s = m'(m)$ as stated in (5.10)

Validation: $N_S \times p_S = m'(m)$

$$\Rightarrow N_S \times p_S = total \quad number \quad of \quad measured \quad idle \quad channels \tag{5.11}$$

where N_s=total number of SUs and p_s= SU arrival probability.

In the ideal scenario, the total number of idle channels is expressed as,

$$m = M\left(1 - p_r\right) \tag{5.12}$$

where M = total number of channels in the CRN and p_r = PU arrival probability.

Therefore, $$N_s \times p_s = m \tag{5.13}$$

Considering $M = N_p$ where each PU is assigned a single channel,

$$m = N_p\left(1 - p_r\right) \tag{5.14}$$

where N_p=total number of PUs.

Substituting the value of m from (5.14) in (5.13),

$$N_s \times p_s = N_p\ \left(1 - p_r\right) \tag{5.15}$$

For identical transmission probabilities of both PU and SU as denoted by p, (5.15) is modified as,

$$N_s \times p = N_p\left(1 - p\right) \qquad \Rightarrow N_s = \frac{N_p\left(1 - p\right)}{p} \tag{5.16}$$

Equation (5.16) has already been established in [5.23]. Since (5.16) is derived from (5.11), (5.11) is validated with respect to [5.22].

Further, it is inferred from Fig. 5.4 that in case of CRN_RESERV, the trade-off is optimal under the condition given by,

$$N_s \times p_s = n \tag{5.17}$$

It is thus confirmed from (5.10) and (5.17) that the optimal number of SUs supported by CRN_RESERV decreases as compared to CRN_UNRESERV, thereby decreasing the overall SU Sum Goodput in CRN_RESERV.

In order to find the optimal number of channels to be reserved for the PUs in CRN_RESERV, the normalized SU Sum Goodput for CRN_UNRESERV and CRN_RESERV are plotted in Fig. 5.5 with respect to the total number of available idle channels. Different scenarios for CRN_RESERV are considered where each scenario corresponds to a certain number of channels being reserved for the PU. As seen from Fig. 5.5, the SU Sum Goodput for CRN_UNRESERV records a steep decline with the decrease in the number of idle channels and thus, can be expressed as a function of PU activity.

Therefore,

$$C_{UNRESERV}^{sum} = f\left(H_i\right) \forall\, i \in N_p \tag{5.18}$$

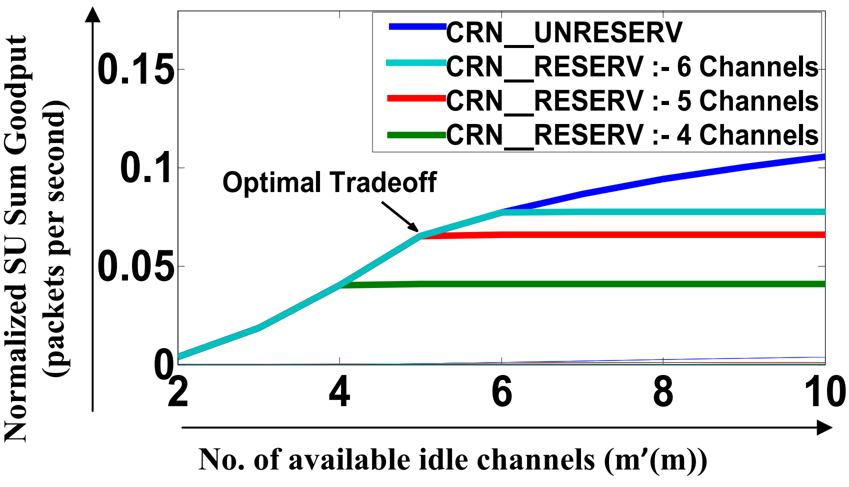


Fig. 5.5 Variation in the Normalized SU Sum Goodput in CRN_UNRESERV and CRN_RESERV with the total number of idle channels for different instances of channel reservation for the PU

CRN_RESERV, on the other hand, remains independent of any PU interference till all the reserved channels are occupied by the PUs, as observed from the fairly constant nature of C_{RESERV}^{sum} in Fig. 5.5. Thus, the SU dropping and handoff probabilities decrease, resulting in overall QoS enhancement for the SU applications. However, with the lesser number of idle channels available to the SUs due to channel reservation, C_{RESERV}^{sum} is less than $C_{UNRESERV}^{sum}$. As denoted in Fig. 5.5, the tradeoff between C_{RESERV}^{sum} and instances of uninterrupted SU transmission is optimal when their product is maximum as given by the following condition.

$$m'(m) - n = \frac{m'(m)}{2} \tag{5.19}$$

Equation (5.19) can also be validated using the fundamental mathematical logic and the proof is provided below.

Proof: Product of the channel utilization and the instances of SU independence from any PU interference is maximum when $m'(m)$-n = $m'(m)/2$

Proof: Let the CRN comprises of m number of idle channels where m-n number of idle channels are reserved for the PUs. Therefore, the total number of idle channels accessed by the SUs is given by,

$$m_{SU} = n \times p_{SU} \tag{5.20}$$

where p_{SU} denotes the probability of an idle channel access by a SU.

The number of idle channels that a PU can access without affecting the existing SU transmissions is expressed as,

$$m_{PU} = m - m_{SU} \tag{5.21}$$

The higher the value of m_{PU}, the more the SU is free from any PU interference. Thus, m_{PU} is a measure of uninterrupted SU transmission. Hence, the product of SU channel utilization and uninterrupted transmission is given by,

$$Prod = m_{SU} \times m_{PU} \tag{5.22}$$

Expressing *Prod* as per (5.20) and (5.21),

$$Prod = n \times p_{SU} \times (m - n \times p_{SU}) \tag{5.23}$$

The objective is to find the optimal value of m-n such that *Prod* is maximum. Differentiating *Prod* with respect to n and equating it to 0,

$$\frac{dProd}{dn} = \frac{d\{n \times p_{SU} \times (m - n \times p_{SU})\}}{dn} = 0$$

$$\Rightarrow n = \frac{m}{2 p_{SU}} \tag{5.24}$$

Taking the second order derivative of *Prod* with respect to *n*,

$$\frac{d^2 Prod}{dn^2} = -2p_{SU}^2 < 0 \tag{5.25}$$

Therefore, *Prod* is maximum when

$$n = \frac{m}{2p_{SU}} \tag{5.26}$$

Considering $p_{SU} = 1$ for all the SUs (denoting channel access by a SU with absolute certainty) and $m'(m)$ to be the total number of measured idle channels, (5.26) is modified as,

$$n = \frac{m'(m)}{2}$$

$$\Rightarrow m'(m) - n = m'(m) - \frac{m'(m)}{2} = \frac{m'(m)}{2} \tag{5.27}$$

Thus, from (5.24), (5.25) and (5.27), it is proved that *Prod* is maximum under the condition given by,

$$m'(m) - n = \frac{m'(m)}{2} \tag{5.28}$$

Thus, the optimal number of channels to be reserved for a PU in CRN_RESERV is given by (5.19).

5.4.5 Comparative Performance Evaluation with the Existing Works in Literature

It is already mentioned in Section 5.1 that all the existing works in literature with respect to PU based channel reservation in the CRN have examined only the basic system parameters and hence, do not provide a complete study of the network performance. Moreover, the previous works have primarily relied on Markov Models for evaluating system performance. As the complexities of the model grows with increasing number of channels, it is imperative that these works have abstained from depicting the important system metrics (blocking, dropping, transmission probabilities) under different number of reserved and unreserved channels.

On the other hand, this chapter addresses this shortcoming by designing robust mathematical models for both CRN_UNRESERV and CRN_RESERV, which provide a complete analysis of all the critical system parameters and enables us to study the system metrics with variation in the number of channels reserved. Therefore, in this section, the completeness and efficiency of the designed analytical framework in this chapter is established with respect to the existing works in this domain. Comparative performance evaluation is done specifically with respect to every single published work in this domain by highlighting their limitations and addressing them effectively in the proposed model.

Considering equally likely channel access by a SU in the CRN, the probability of a SU successfully occupying a channel from s number of channels is derived from (5.9) and is given by,

$$P_{SU\_Tx} = (1 - p_r)\binom{s}{1}\left(\frac{1}{s}\right)\left(1 - \frac{1}{s}\right)^{N_s p_s - 1} \tag{5.29}$$

where p_r = probability of PU arrival in that channel, p_s = SU arrival probability and N_s = total number of SUs.

The different SU probabilities such as the blocking probability, dropping probability, handoff probability and the probability of successful transmission are derived from (5.29) and thoroughly examined with respect to the previous works.

The blocking probability (p_b) is defined as the probability that an incoming user cannot be accepted in the network. The designed Markov model in [5.15] has mapped p_b for a SU with respect to the SU arrival rate only and has not investigated the rest of the parameters. On the contrary, based on the analytical framework for CRN_UNRESERV and CRN_RESERV in this chapter, the SU blocking probabilities are examined with respect to three parameters, namely i) the total number of SUs in CRN in Fig. 5.6, ii) SU arrival probability in Fig. 5.7, and iii) the total number of available idle channels in Fig. 5.8. The effect of reserving different channels for the PUs on p_b is also analyzed for all these three scenarios.

It is observed from Fig. 5.6 that p_b increases with a rise in the total number of SUs in the CRN and is higher for CRN_RESERV than CRN_UNRESERV, and the obtained pattern is similar to the output as demonstrated in [5.15].

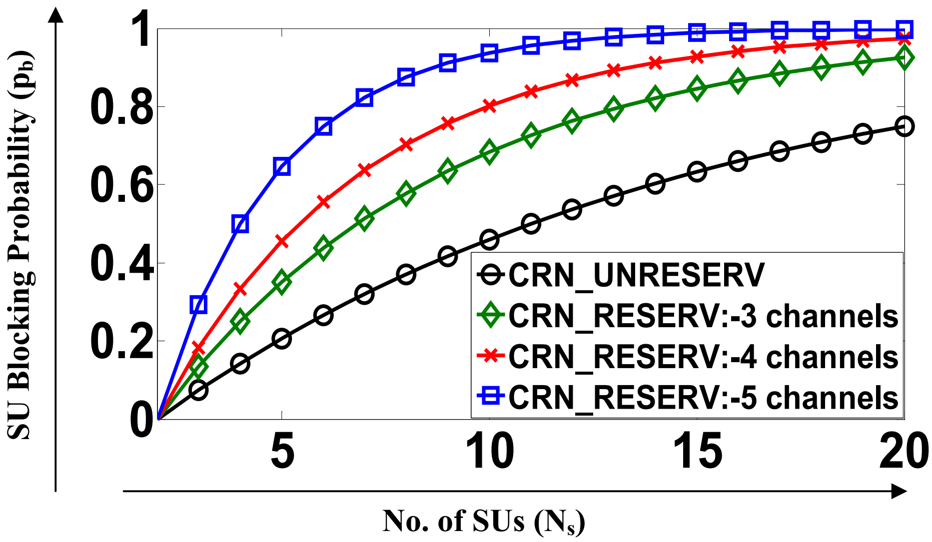


Fig. 5.6 Variation in the SU Blocking Probability in CRN_ UNRESERV and CRN_RESERV for different number of reserved channels with the total no. of SUs

In addition, as the probability of the SU arrival in the CRN increases, higher number of SUs competes for the idle channels, leading to an increase in p_b as shown in Fig. 5.7.

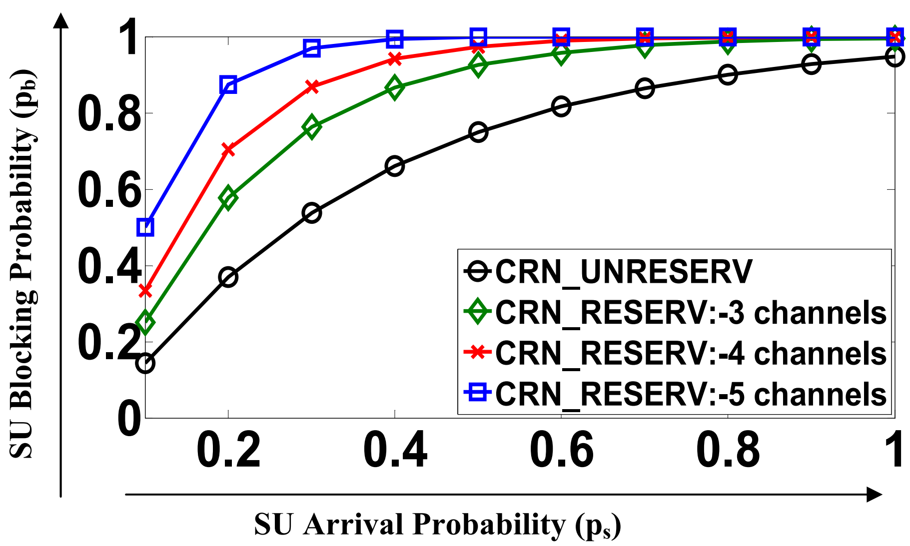


Fig. 5.7 Variation in the SU Blocking Probability in CRN_ UNRESERV and CRN_RESERV for different number of reserved channels with SU Arrival Probability

It is also evident in both Fig. 5.6 and Fig. 5.7 that p_b increases as more number of idle channels are reserved for the PUs.

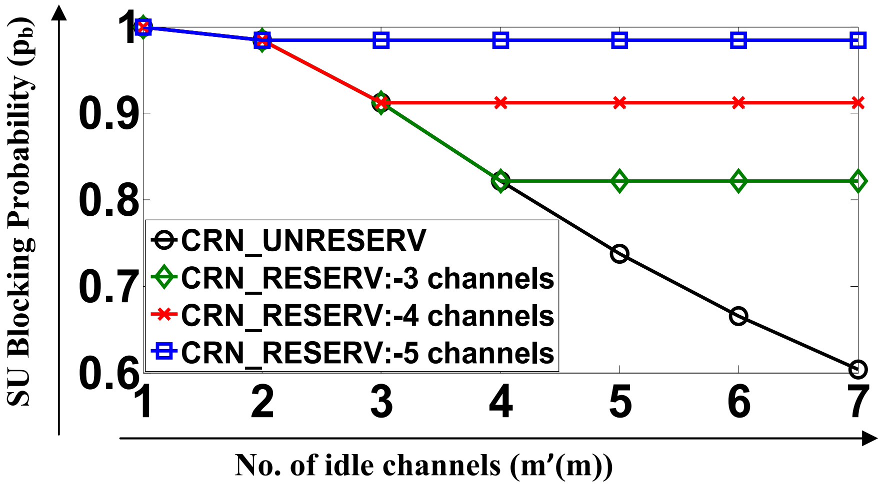


Fig. 5.8 Effect of the no. of idle channels on SU Blocking Probability

The effect of the different number of reserved channels on p_b as depicted in [5.17] and [5.18] is also addressed in this work. While [5.17] and [5.18] have analyzed the variation in p_b with the number of reserved channels only, the model in this work facilitates the study of p_b in terms of both the number of reserved channels and the total number of idle channels. It is shown in Fig 5.8 that p_b decreases as the number of reserved channels is reduced, thereby confirming the output from [5.17] and [5.18]. Further, p_b decreases with an increase in the number of available idle channels for the SUs. However, p_b becomes constant after a certain time period in CRN_RESERV because as per the policy of CRN_RESERV, these SUs cannot access the remaining idle channels that are marked reserved for the PUs.

Next, we examine the dropping probability which is defined as the probability that the user is forced to stop its activity and vacate the channel. The SU dropping probability (p_{dp}) has been studied with respect to the SU arrival rate in [5.15] and the PU arrival rate in [5.16]. On the contrary, this chapter maps the SU dropping probability with the total number of available idle channels as shown in Fig. 5.9. The advantage of this plot is that it further incorporates the variation with different number of reserved channels in a CRN. It is observed from Fig. 5.9 that p_{dp} decreases as more number of idle channels is available for the SUs. Moreover, p_{dp} confirms the outcome in [5.15] and records a decrease in CRN_RESERV compared to CRN_UNRESERV. As the number of channels to be reserved for the PUs is increased, p_{dp} also decreases for a particular SU as seen from Fig 5.9, thereby assuring longer periods of interruption-free transmission for the SU.

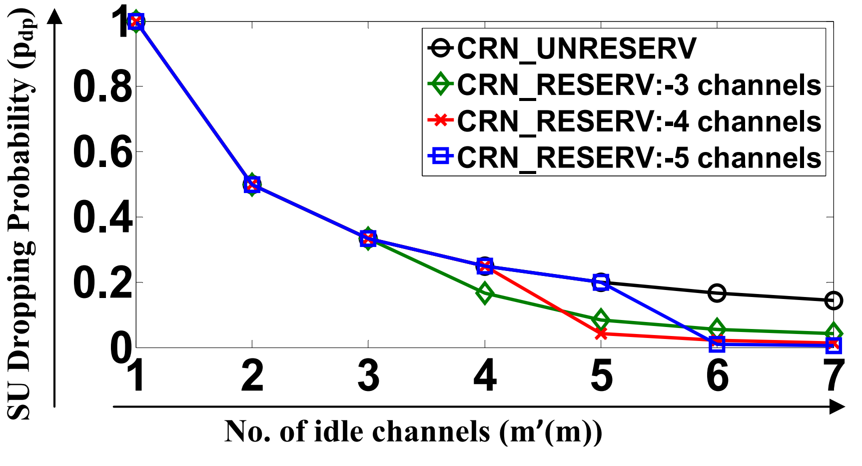


Fig. 5.9 Effect of the no. of idle channels on SU Dropping Probability

The plot in Fig 5.9 also leads to the study of successful SU transmission probability (p_{tx}) with variation in the PU arrival probability in Fig. 5.10 that has not been studied in any of the previous works. It must be noted in this regard that the PU arrival probability (p_r) is different from the PU transmission probability. Rise in p_r triggers the arrival of more PUs in the CRN, leading to increased instances of the SUs vacating their channels. Hence, p_{tx} decreases with the rise in p_r. With further increase in the number of reserved channels in CRN_RESERV, p_{tx} decreases as the probability of a PU occupying an unreserved channel decreases.

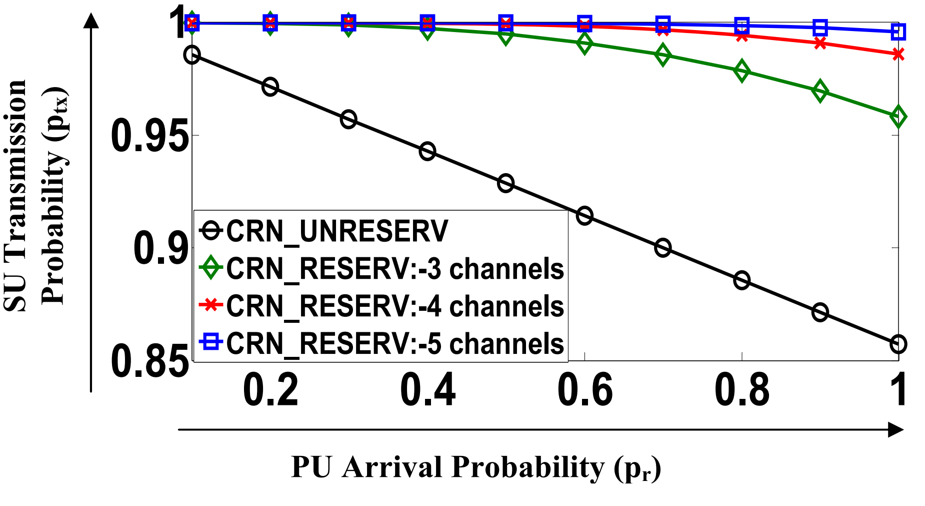


Fig. 5.10 Variation in the SU Transmission Probability with the PU Arrival Probability

The SU handoff probability (p_{hd}) is defined as the probability with which a SU performs spectrum handoff to an idle channel on any PU arrival on the current channel. This concept of spectrum handoff has been incorporated in the Markov Model in [5.16] and both the dropping and blocking probabilities are plotted for CRN_RESERV and CRN_UNRESERV. However, the variation of p_{hd} has not been studied in [5.16]. Hence, this work maps p_{hd} with respect to the PU activity and the total number of channels reserved for the PUs.

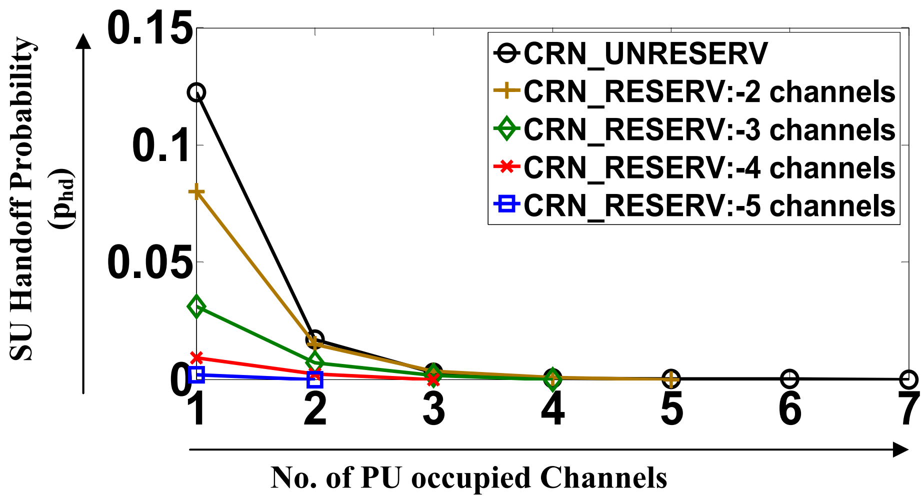


Fig. 5.11 Variation in the SU Handoff Probability with the no. of PU occupied channels

It is seen from Fig. 5.11 that p_{hd} decreases as more channels are occupied by the PUs. Moreover, p_{hd} is high in CRN_UNRESERV as higher number of channels are available to a particular SU for performing spectrum handoff, than CRN_RESERV. It is also recorded that p_{hd} decreases with an increase in the number of reserved channels.

Finally, the PU behavior in CRN_RESERV is illustrated in [5.18] in terms of the PU blocking probability. Comparatively, the mathematical model for CRN_RESERV in this chapter plots not only the PU blocking probability (p_{rb}) but also their probability of successful channel access (p_{ch_acc}).

A decrease in p_{rb} with the increase in the number of reserved channels as recorded in [5.18] is confirmed in Fig 5.12. This also validates the findings of our model with that of the published works in literature.

It is further observed that p_{rb} increases as more number of PUs arrives in the CRN.

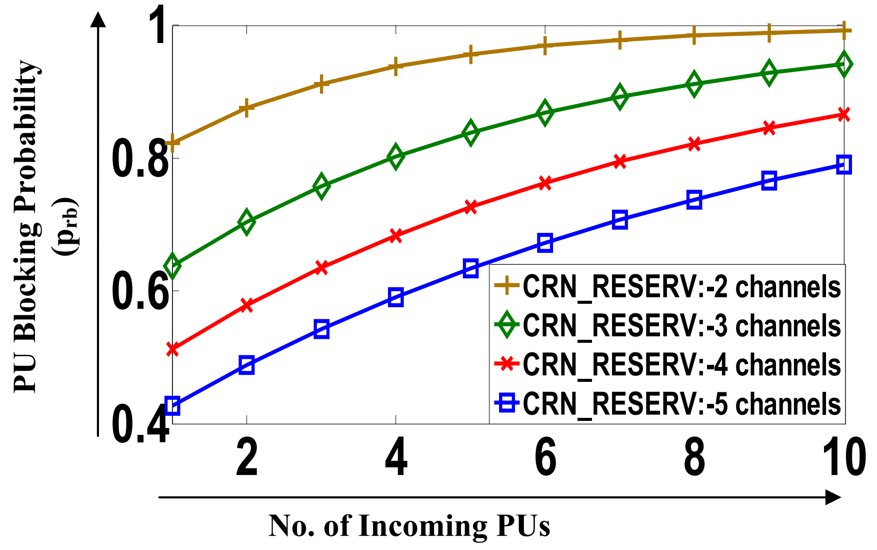


Fig. 5.12 PU Blocking Probability vs. the total no. of incoming PUs

Fig 5.13 plots the probability of successful channel access (p_{ch_acc}) by a PU with respect to the PU arrival probability (p_r) in the unreserved channels of CRN_RESERV. As p_r increases, more PUs compete for the channel, resulting in the decline in p_{ch_acc}. An interesting aspect of Fig. 5.13 is that with an increase in the number of reserved channels for the PU, p_{ch_acc} for the unreserved channels decreases, thereby reducing the interference between the PU and SU transmissions.

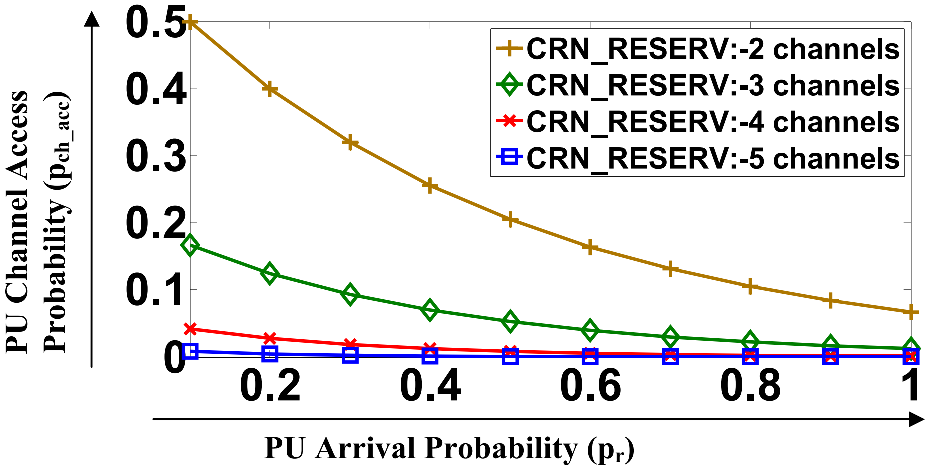


Fig. 5.13 Variation in PU Channel Access Probability with the PU Arrival Probability

Thus, the developed mathematical model provides a complete study of all the SU and PU probabilities and their variation with the different critical system parameters, in contrast to the earlier works (that provide only a partial overview of CRN_RESERV and CRN_UNRESERV) and thereby establishes the novel contributions of this work as stated in Table 5.1.

5.5 Implementing Channel Reservation Policy for VoIP Applications

From the previous section, it is deduced that CRN_RESERV provides higher QoS (in terms of low dropping and handoff probabilities and higher transmission probability) at the cost of a decrease in the system throughput as compared to CRN_UNRESERV. This makes it suitable for the real-time applications that demand stringent QoS requirements. In this regard, this chapter implements the channel reservation strategy in a VoIP based CRN [5.21] to study its effect on the VoIP call quality. The success of a standard VoIP communication depends on several quality metrics namely, delay, jitter, packet loss, etc. [5.23] that should be minimized and maintained below the threshold level. However, unlike the traditional wireless networks, the CRN introduces a new delay called *spectrum handoff delay* for the VoIP users. The handoff delay of a spectrum handoff is defined as the duration from the time a SU vacates the current channel to the time it resumes the transmission after accessing an idle channel successfully [5.20]. This delay component increases the total delay incurred in a VoIP transmission and can adversely affect the call quality when the overall end-to-end delay rises above 180 ms. Moreover, the probability of interference with the PUs increases, resulting in the packets being rendered erroneous and considered lost. As VoIP communication is based on the successful delivery of the IP packets, a packet loss of greater than 5-6% significantly disrupts the speech quality [5.23]. Therefore, in order to reduce this additional delay and packet loss, the channel reservation policy is implemented in a VoIP based CRN.

5.5.1 Formulation of the VoIP traffic

VoIP communication occurs in the form of talkspurts [5.20]. Hence, the SU transmits during the talkspurt period and remains idle during the silent suppression period with probabilities P_{busy} and P_{idle}, respectively. Accordingly, the VoIP traffic for a single SU is formulated as an on-off model with $1/\alpha$ and

$1/\beta$ as on and off periods, respectively, that are exponentially distributed [5.21]. Therefore, the probability with which the SU is busy transmitting is given by,

$$P_{busy} = \frac{\alpha^{-1}}{\alpha^{-1} + \beta^{-1}} \quad (5.30)$$

Let $R_q(V)$ be the VoIP transmission rate for a SU where q denotes an idle channel and V represents the VoIP transmission. It is worth mentioning that the average VoIP transmission rate is 50 or 33.3 packets per second and depends on the codecs implemented [5.23]. The corresponding throughput of the SU in t_d time at a particular idle channel is expressed as,

$$C_{SU} = R_q(V) \times t_d \times P_{busy} \quad (5.31)$$

5.5.2 Design of CRN_UNRESERV for the VoIP SU

Let the CRN consists of m idle available channels as per (5.1) to be allotted to N_p number of PUs. It is considered that each PU is active with a probability p_r. Therefore, the probability of occupying a single channel by a PU is given by the following expression.

$$P_{pr\_occ}^{unreserv} = \binom{m}{1} \frac{1}{m} \left(1 - \frac{1}{m}\right)^{N_p \times p_r - 1} \quad (5.32)$$

As the PU arrival varies with time, (5.32) is modified with respect to a particular time interval t and is given by,

$$P_{pr\_occ}^{unreserv}(t) = \left(1 - \frac{1}{m}\right)^{N_p \times p_r(t) - 1} \quad (5.33)$$

Considering the channel allocation as an independent event, the probability that all of the m channels are occupied by the PUs is given by $P_{pr\_occ}(t)^m$. Therefore, at least a single channel is idle at time t with the probability as expressed below.

$$P_{channel\_idle}^{unreserv}(t) = 1 - P_{pr\_occ}^{unreserv}(t)^m = 1 - \left(1 - \frac{1}{m}\right)^{m\left(N_p \times p_r(t) - 1\right)} \quad (5.34)$$

Every VoIP talkspurt for a SU consists of an on and off period, denoted by t_{on} and t_{off}, respectively. When a PU arrives in the *on* state of a VoIP call, interference occurs until the SU suspends its transmission and performs spectrum handoff. Let $t_{inf}(i)$ be the time for which this interference occurs in the ith talkspurt. Moreover, let $d_{handoff}$ be the mean spectrum handoff delay associated with each instance of spectrum handoff. This delay depends on the spectrum handoff strategy selected by the SU [5.24]. It is obvious that the SU performs handoff to an idle channel only if it is available, that, in turn, is governed by the dropping probability in the CRN. Therefore, a VoIP talkspurt for the ith on-off cycle can be expressed as,

$$T_{talkspurt}(i) = \left\{ \overset{\text{No PU Presence}}{1 - P_{pr\_occ}^{unreserv}(t_i)} \right\}\left\{ \overset{\text{VoIP talkspurt}}{t_{on}(i) + t_{off}(i)} \right\} + \underset{\text{PU Presence}}{P_{pr\_occ}^{unreserv}(t_i)}\left\{ t_{on}(i) - t_{inf}(i) + \underset{\text{Handoff}}{P_{channel\_idle}^{unreserv}(i)} \times d_{handoff} \right\} \tag{5.35}$$

The total time of interference and the total handoff delay during an entire VoIP communication comprising of g talkspurts are derived using (5.33), (5.34) and (5.35) and are expressed as follows.

$$T_{total\_inf}^{unreserv} = \sum_{i=1}^{g}\left\{ \left(1 - \frac{1}{m}\right)^{N_p \times p_r(i) - 1} \times t_{inf}(i) \right\} \tag{5.36}$$

$$T_{total\_handoff}^{unreserv} = \sum_{i=1}^{g-1}\left\{ \left(1 - \frac{1}{m}\right)^{N_p \times p_r(i) - 1} \times \left\{ 1 - \left(1 - \frac{1}{m}\right)^{m\{N_p \times p_r(i) - 1\}} \right\} \right\} \times d_{handoff} \tag{5.37}$$

5.5.3 Design of CRN_RESERV for the VoIP SU

In this scenario, it is considered that CRN_RESERV reserves (m-n) number of channels for the PUs, thereby allowing the SUs to access n number of idle channels. The probability of occupying a single channel by a PU in

CRN_RESERV at time interval t is derived by modifying (5.33) and is expressed as,

$$P^{reserv}_{pr\_occ\_reserv}(t) = \left(1 - \frac{1}{m-n}\right)^{N_p \times p_r(t) - 1} \tag{5.38}$$

A PU can access the unreserved channels only when all the reserved channels are occupied by other PUs. The probability of accessing an unreserved idle channel by a PU is expressed as,

$P^{reserv}_{pr\_occ\_unreserv}(t) = Probability\ that\ all\ (m-n)\ channels\ are\ occupied$

$\times\ Probabilit\ y\ of\ occupying\ a\ single\ channel\ from\ n\ channels$

$$= \left(P^{reserv}_{pr\_occ\_reserv}(t)\right)^{(m-n)} \times \left(1 - \frac{1}{n}\right)^{\{N_p - (m-n)\}p_r(t) - 1} \tag{5.39}$$

The probability that all the unreserved channels are occupied is given by $\left(P^{reserv}_{pr\_occ\_unreserv}(t)\right)^n$. Hence, the probability that at least one channel is idle for SU transmission is provided by the following expression.

$$P^{reserv}_{channel\_idle}(t) = 1 - \left(P^{reserv}_{pr\_occ\_unreserv}(t)\right)^n$$

$$= 1 - \left[\left(1 - \frac{1}{m-n}\right)^{(N_p \times p_r(t) - 1)(m-n)n} \times \left(1 - \frac{1}{n}\right)^{n[\{N_p - (m-n)\}p_r(t) - 1]}\right] \tag{5.40}$$

Accordingly, (5.36) and (5.37) are modified to derive the total time of interference and handoff delay, respectively for the VoIP SUs in CRN_RESERV and are expressed as follows.

$$T^{reserv}_{total\_inf} = \sum_{i=1}^{g}\left\{\left(1 - \frac{1}{m-n}\right)^{\{N_p \times p_r(i) - 1\}(m-n)} \times \left(1 - \frac{1}{n}\right)^{\{N_p - (m-n)\}p_r(i) - 1} \times t_{inf}(i)\right\} \tag{5.41}$$

$$T_{total\_handoff}^{reserv} = \sum_{i=1}^{g-1}\left\{\left(1-\frac{1}{m-n}\right)^{\{N_p \times p_r(i)-1\}(m-n)} \times \left(1-\frac{1}{n}\right)^{\{N_p-(m-n)\}p_r(i)-1} \times \left\{1-\left\{\left(1-\frac{1}{m-n}\right)^{\{N_p \times p_r(i)-1\}(m-n)n} \times \left(1-\frac{1}{n}\right)^{[\{N_p-(m-n)\}p_r(i)-1]n}\right\}\right\}\right\} \times d_{handoff}$$

(5.42)

It is to be noted that the interference time in (5.41) is governed by the PU arrival probability in the unreserved channels as per (5.39). In addition, the handoff delay as expressed in (5.42) depends on the spectrum handoff instances which can occur only when there is atleast one idle channel in the system (as governed by (5.40)) and the same is incorporated in (5.42). It is evident from (5.41) and (5.42) that only when PU arrives in the unreserved channels, interference and spectrum handoff conditions arise for the SUs.

Thus, the most significant performance parameters for the VoIP calls in the CRN are derived under the consideration that all the SUs arrive with an equal probability. Thereafter, the channel allocation for a SU is considered in a multiple user scenario, where every SU arrives with different transmission probability. Under such circumstances, the SUs access different idle channels with different probabilities. Let $p_s(t)$ be the SU arrival probability at time interval t. From the SU perspective, there are $m'(m)$ number of idle channels considering the effect of false alarm and miss detection as already derived in (5.2). Therefore, the probability of accessing an idle channel by a single SU among N_s SUs is given by the expression as follows.

$$P_{sec\_occ}(t) = \left(1-\frac{1}{m'(m)}\right)^{N_s \times p_s(t)-1} \tag{5.43}$$

The channel access by the SUs with respect to different SU and PU arrival probabilities in CRN_RESERV is represented in the form of a tree in Fig. 5.14. Every branch of the tree provides the time for which a SU accesses the channel corresponding to a particular VoIP talkspurt. It is obvious from the figure that as long as the reserved channels are idle for the PUs to access, the

SUs enjoy uninterrupted transmission. Once the reserved channels are occupied, two conditions arise with respect to the PU arrival on the unreserved channels and are described as follows.

1. As long as a PU abstains from accessing a channel used for transmission by a SU, the SU continues its VoIP transmission without any issue and is represented by the left node of every sub-tree in Fig. 5.14. The best-case scenario occurs when the SU completes its VoIP communication without facing any interruption due to the incoming PUs and is represented by the leftmost leaf node of the tree in Fig. 5.14. The total time of transmission by the SU in this scenario is given by,

$$T_{SUtx\_bestcase}^{reserv} = \left(1-\frac{1}{n}\right)^{N_s \times p_s(1)-1} \sum_{i=1}^{g}\left[\prod_{k=1}^{i}\left\{1-\left(1-\frac{1}{m-n}\right)^{\{N_p \times p_r(k)-1\}(m-n)} \times\left(1-\frac{1}{n}\right)^{\{N_p-(m-n)\}p_r(k)-1}\right\}\times t_{on}(i)\right] \tag{5.44}$$

2. Once a PU arrives in the channel used by the SU, interference occurs and the SU performs spectrum handoff to another available channel, as illustrated by the right node of every sub-tree in Fig. 5.14. It is considered that all the reserved channels are occupied by the PUs by the time the SU completes its xth talkspurt period. The worst-case scenario arises when a PU arrives at every talkspurt, resulting in interference and triggering spectrum handoff and corresponds to the rightmost leaf node of the tree. The SU transmission time in this scenario is derived as follows.

$$\begin{aligned} T_{SUtx\_worstcase}^{reserv} = \left(1-\frac{1}{n}\right)^{N_s \times p_s(1)-1} & \sum_{i=1}^{x}\left[\prod_{k=1}^{i}\left\{\left(1-\frac{1}{m-n}\right)^{\{N_p \times p_r(k)-1\}(m-n)} \times\left(1-\frac{1}{n}\right)^{\{N_p-(m-n)\}p_r(k)-1}\right\}\times t_{on}(i)\right] \\ & + \sum_{i=x+1}^{g}\left[\prod_{k=1}^{i}\left\{\left(1-\frac{1}{m-n}\right)^{\{N_p \times p_r(k)-1\}(m-n)} \times\left(1-\frac{1}{n}\right)^{\{N_p-(m-n)\}p_r(k)-1}\right\} \right. \\ & \left. \times\left(1-\frac{1}{n-i+1}\right)^{N_s \times p_s(i)-1} \times\left\{t_{on}(i)-t_{inf}(i)\right\}\right] \end{aligned} \tag{5.45}$$

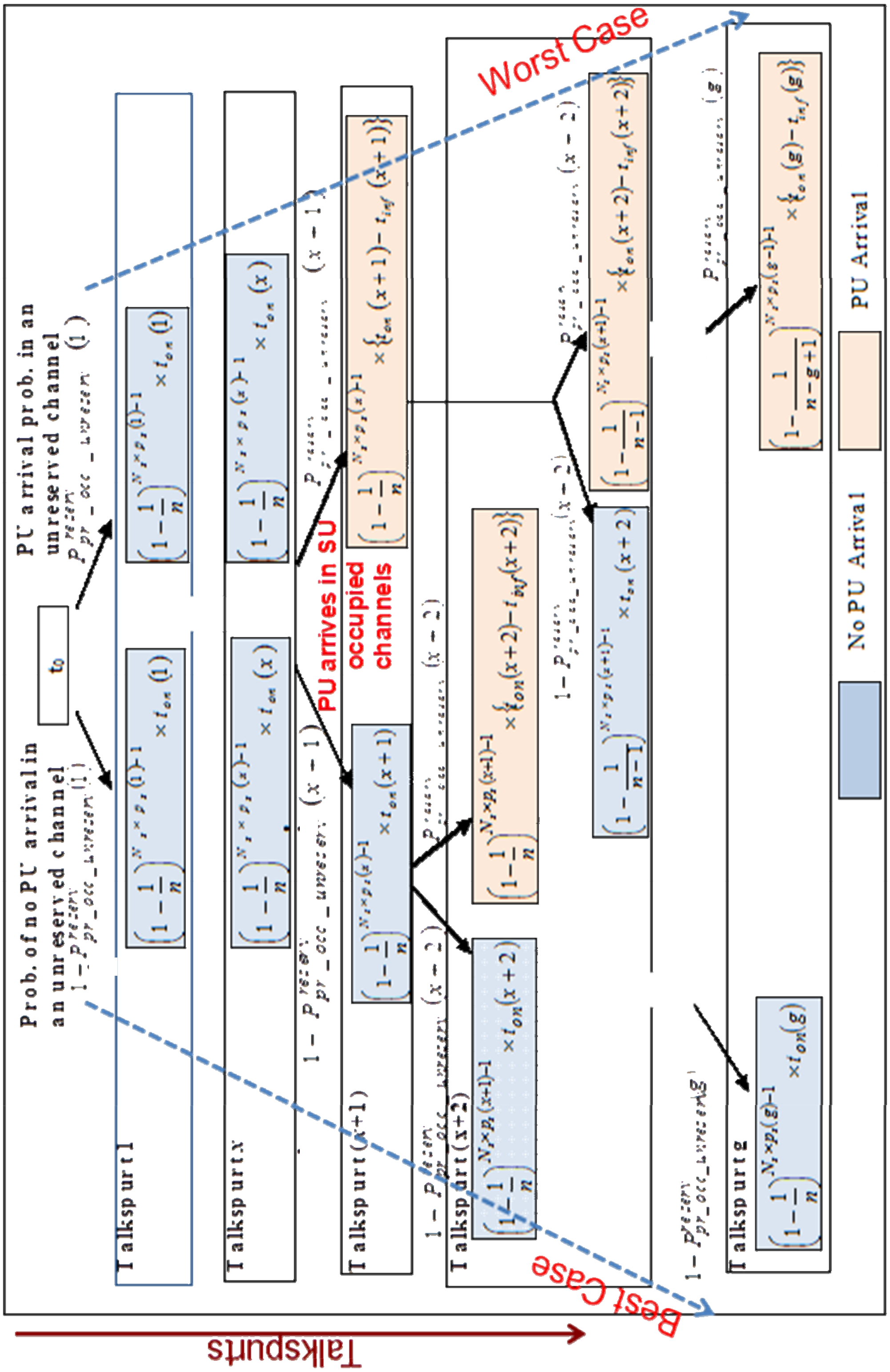


Fig. 5.14 Representation of channel access by the VoIP SUs with respect to different SU and PU arrival probabilities in CRN_RESERV

5.6 Priority based Adaptive Variation in Channel Reservation: *PACR* Algorithm

There are two drawbacks of CRN_RESERV. Firstly, it implements the static channel reservation policy where a fixed number of channels is reserved for PUs, leading to idle channels being inaccessible to SUs during low PU activity. Secondly, in the absence of any QoS provisioning mechanism for the SUs, once all the reserved channels are occupied, the incoming PUs can access any of the unreserved channels with equal probability, leading to significant call drops for SUs. Both these problems are highlighted in the following figure.

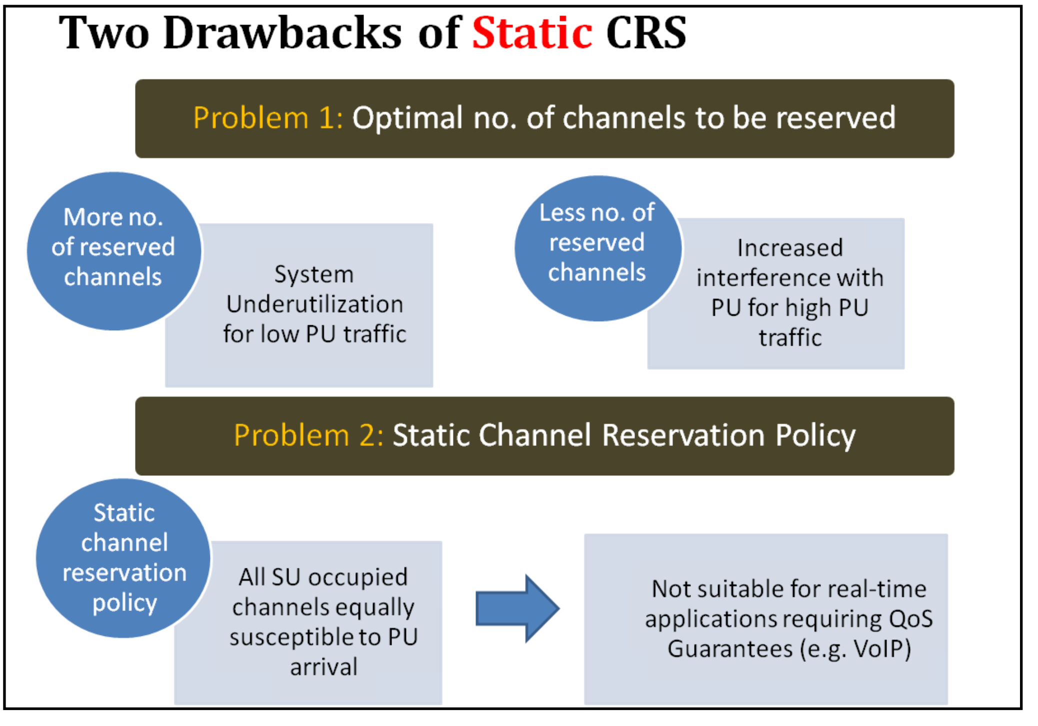


Fig. 5.15 Drawbacks of Static CRS (Channel Reservation Scheme)

Therefore, in order to address these limitations, *Priority based Adaptive Channel Reservation* (*PACR*) algorithm is designed in this section. In addition to the advantages provided by the channel reservation policy (as already described in the previous sections), the proposed algorithm further improves channel utilization by incorporating adaptive channel reservation strategy based on PU activity and allows priority based access to different categories of SU traffic, leading to increase in system capacity and heterogeneity.

5.6.1 Design of *PACR* Algorithm

PACR algorithm comprises of two major processes and is outlined below. A centralized CRN is considered where spectrum broker [5.20] coordinates between VoIP and data SUs.

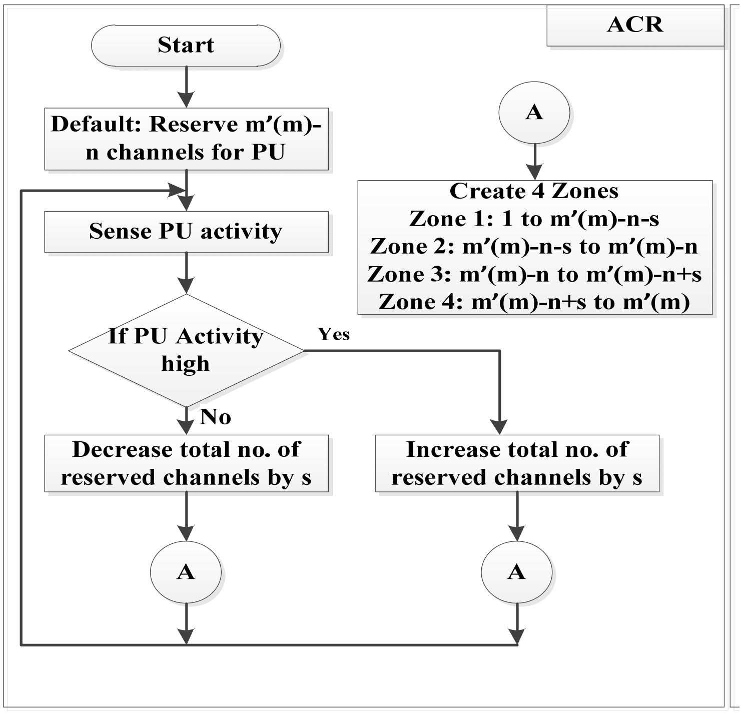
(i) Adaptive Channel Reservation (ACR)

Adaptive Channel Reservation (*ACR*)	
Step 1.	Default Case: Initially, reserve optimal number of channels for PUs as given by $m'(m)$-n=$m'(m)/2$, where m=total number of idle channels, $m'(m)$= total number of channels sensed idle by SU, n = number of channels accessible by SUs for transmission.
Step 2.	Low PU activity Case: If PU activity is low as decided by spectrum broker, decrease the value of $m'(m)$-n by s.
Step 3.	High PU activity Case: If PU activity is high as decided by spectrum broker, increase the value of $m'(m)$-n by s.

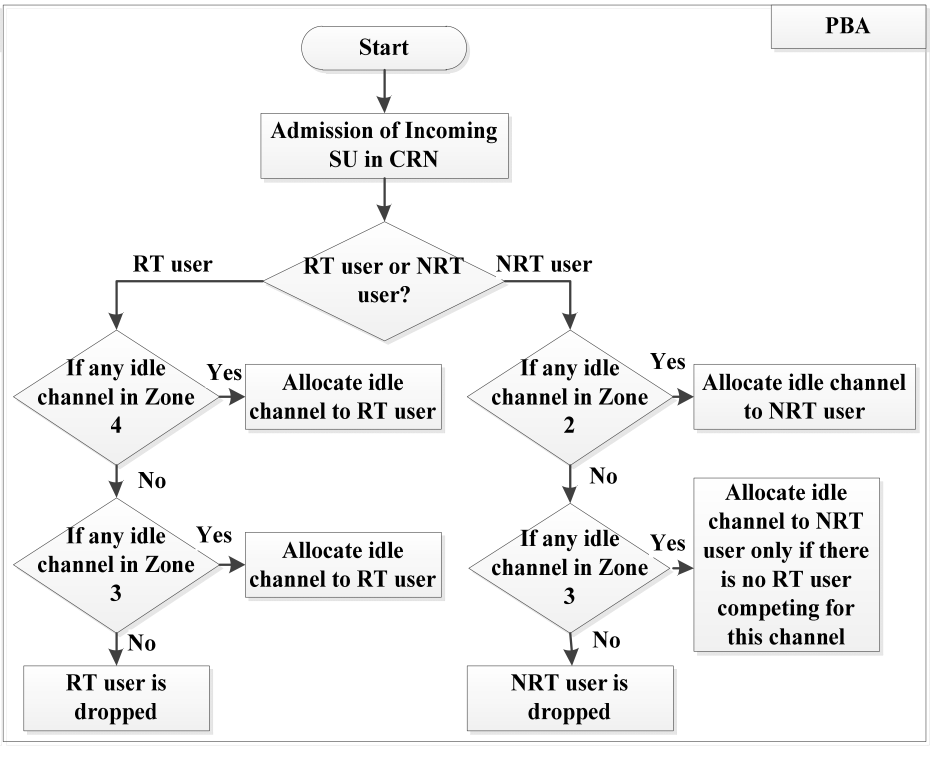
(ii) Priority Based Allocation (PBA)

Priority Based Allocation (*PBA*)	
Step 1.	Categorize SU applications as RT (Real-time) for VoIP users and NRT (Non Real-Time) for data users.
Step 2.	Allow exclusive access of the channels from ($m'(m)$-n+s) to $m'(m)$ to RT users.
Step 3.	Allow exclusive access of the channels from ($m'(m)$-n-s) to ($m'(m)$-n) to NRT users.
Step 4.	Allow access of channels from ($m'(m)$-n) to ($m'(m)$-n+s) to both RT and NRT users, with RT users having higher priority for channel access than NRT users.

The flowchart for *PACR* algorithm is depicted in Fig. 5.16



(a)



(b)

Fig. 5.16 Flowchart illustrating the proposed *PACR* Algorithm

5.6.2 Discussion of the Algorithm

ACR divides the entire range of available channels into four zones. Initially, the spectrum broker reserves optimal number of channels for PUs, as derived in (5.19). However, if it finds the number of PU occupied channels less than optimal value (resembling low PU activity), it reduces the number of reserved channels for PU by *s*, creating zone 1. Zone 1, thus, comprises of channel 1 to channel (*m′(m)-n-s*). As soon as PUs occupy all channels in zone 1, spectrum broker further reserves an additional *s* number of channels for PU. This region from channel *m′(m)-n-s* to channel *m′(m)-n* constitutes zone 2. Zone 3 comprises of channel *m′(m)-n* to channel *m′(m)-n+s*, that are again reserved for PU when all the previous channels are occupied (denoting high PU activity). Finally, the remaining number of channels from *m′(m)-n+s* to *m′(m)* form zone 4. Thus, *ACR* adaptively reserves number of channels for PUs based on their activities and is described in the following figure.

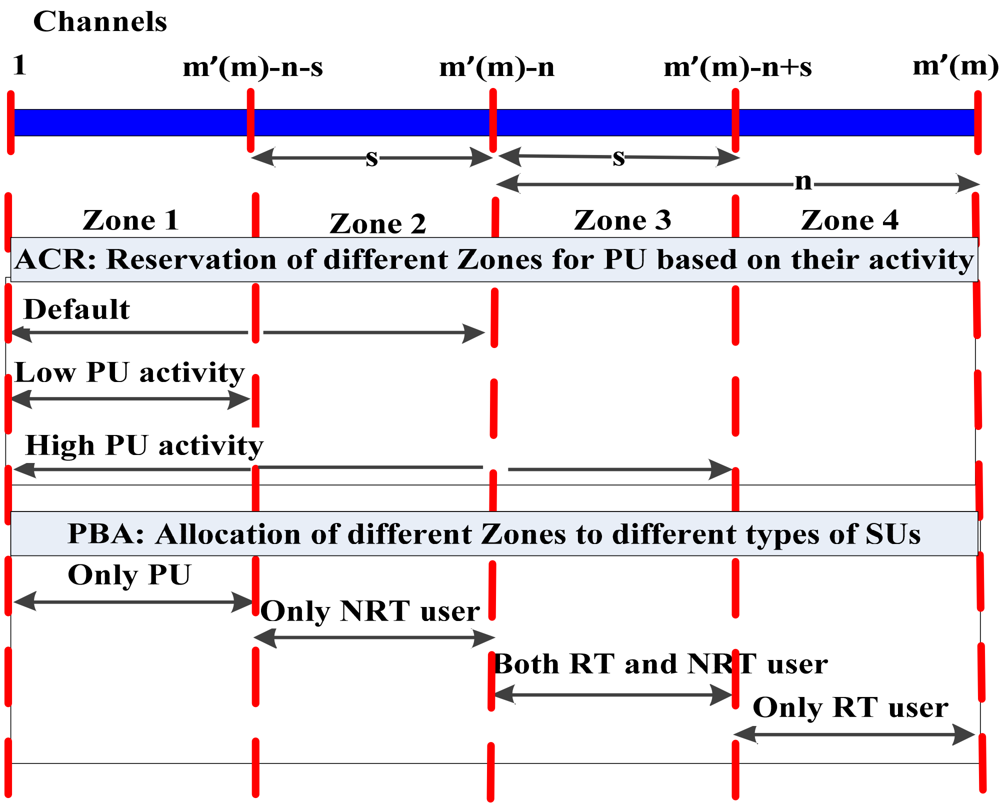


Fig. 5.17 Schematic Diagram depicting the functions of *ACR* and *PBA* in *PACR* algorithm

These 4 zones as created by the *ACR* module are used by the *PBA* module to allocate them to VoIP and data users, as denoted by RT and NRT users respectively. The aim is to provide maximum QoS protection and ensure minimal PU interference for RT users. Hence, the channels in zone 4 are kept for exclusive use by RT users only, as this zone is the least susceptible to PU interference compared to other zones. Zone 3 is marked for mutual use by both RT and NRT users. Owing to real-time requirements, RT users enjoy higher priority in channel access than NRT users in zone 3. Zone 2 is the most susceptible to PU interference and is accessed by NRT users during low PU activity. Zone 1 is always exclusively reserved for PUs. It is imperative that while *PBA* ensures maximum QoS protection for RT users in zone 4, it also increases overall utilization of the idle channels by allowing access of additional channels to NRT users in zone 2. PBA is also pictorially described in Fig. 5.17.

5.6.3 Analytical Framework for *PACR* based CRN

The proposed *PACR* algorithm is tested for efficiency by designing a suitable mathematical model followed by rigorous stochastic analysis in this section. The main purpose of this section is to analytically realize the gain in performance through adaptive variation of reserved channels and priority based allocation to different types of SUs as per the *PACR* algorithm. Initially, the underlying mathematical framework is presented based on which important inferences on system performance and SU activity are derived and explained.

It must be mentioned that unlike the previous sections where there were only two cases namely with and without reservation, here we have different reservation scenarios under the effect of *PACR* algorithm. Hence, CRN_UNRESERV is also denoted by CRN_BASIC and both these terms are interchangeable used henceforth.

(i) ***<u>Design and Analysis of Mathematical Model</u>***

The model for *PACR* based CRN is established with respect to SU Sum Goodput. SU Sum Goodput has been derived for CRN_BASIC in (5.7) and subsequently modified for CRN_RESERV in (5.9). This section implements *PACR* in the mathematical model by introducing the parameter “*s*” which defines the adaptive behavior of channel reservation.

Let Ns be the total number of SUs, each transmitting at a rate R_j in the jth idle channel with probability p_s. The throughput for SU per unit time is denoted by C_q. At any instance, the number of available idle channels will belong to any one of the 4 zones as created by *ACR*. The expression for C_q is derived for all the 4 scenarios as follows.

1. *Scenario 1: PU arrives in Zone 1, total number of channels = m′(m), m′(m)-n-s channels reserved for PU, (n+s) channels available for SU.*

$$C_q = H_k \sum_{q=1}^{n+s} \left\{ R_q \binom{n+s}{1} \left(\frac{1}{n+s} \right) \left(1 - \frac{1}{n+s} \right)^{j-1} \right\}$$

$$where\ 1 \le k \le m'(m) - (n+s) \tag{5.46}$$

2. *Scenario 2: PU arrives in Zone 2, total number of channels = m′(m), m′(m)-n channels reserved for PU, n channels available for SU.*

$$C_q = H_k \sum_{q=1}^{n} \left\{ R_q \binom{n}{1} \left(\frac{1}{n} \right) \left(1 - \frac{1}{n} \right)^{j-1} \right\}$$

$$where\ m'(m) - (n+s) + 1 \le k \le m'(m) - n \tag{5.47}$$

3. *Scenario 3: PU arrives in Zone 3, total number of channels = m′(m), m′(m)-n+s channels reserved for PU, (n-s) channels available for SU.*

$$C_q = H_k \sum_{q=1}^{n-s} \left\{ R_q \binom{n-s}{1} \left(\frac{1}{n-s} \right) \left(1 - \frac{1}{n-s} \right)^{j-1} \right\}$$

$$where\ m'(m) - n + 1 \le k \le m'(m) - n + s \tag{5.48}$$

4. *Scenario 4: PU arrives in Zone 4, total number of channels = m′(m), rest of the available channels for PU to occupy, SU occupies any idle channel or else dropped.*

$$C_q = H_k \sum_{q=1}^{m'(m)-k} \left\{ R_q \binom{m'(m)-k}{1} \left(\frac{1}{m'(m)-k} \right) \left(1 - \frac{1}{m'(m)-k} \right)^{j-1} \right\}$$

$$where\ m'(m) - n + s + 1 \le k \le m'(m) - 2 \tag{5.49}$$

Therefore, the SU Sum Goodput for *PACR* based CRN is derived in (5.50) after incorporating all these four scenarios. Considering the value of *s* to be 1, the minimum number of idle channels as denoted by *m′(m)* must be 5 to support *PACR*. The optimal value of *s* depends on several network parameters such as PU and SU arrival rates, their transmission probabilities, total number of available idle channels, etc.

$$C_{PACR}^{sum}(m)=\sum_{m=5}^{Np}\binom{Np}{m}p_r^{\,Np-m}(1-p_r)^m\sum_{j=1}^{Ns}\binom{N_s}{j}p_s^{\,j}(1-p_s)^{Ns-j}\left[\left\{(1-H_1)+\sum_{i=1}^{m'(m)-(n+s)}H_i\right\}\times\right.$$

$$\sum_{q=1}^{n+s}\left\{R_q\binom{n+s}{1}\left(\frac{1}{n+s}\right)\left(1-\frac{1}{n+s}\right)^{j-1}\right\}+\sum_{i=m'(m)-(n+s)+1}^{m'(m)-n}\left\{H_i\left\{\sum_{q=1}^{n}\left\{R_q\binom{n}{1}\left(\frac{1}{n}\right)\left(1-\frac{1}{n}\right)^{j-1}\right\}\right\}\right\}$$

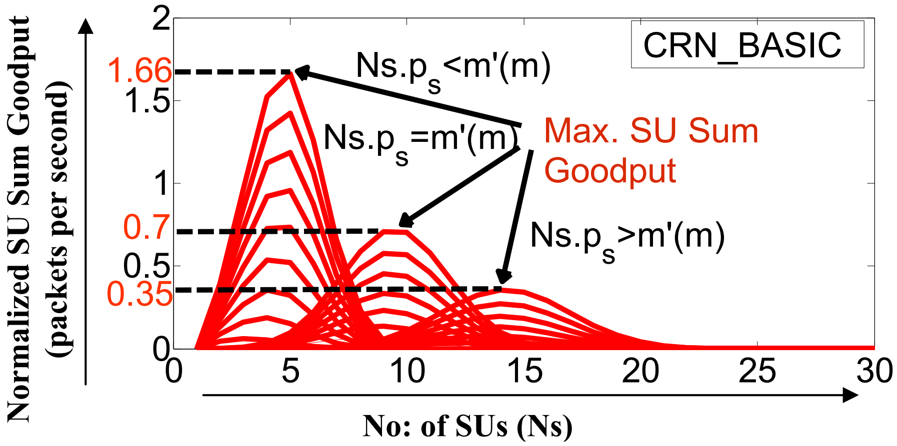
$$+\sum_{i=m'(m)-n+1}^{m'(m)-n+s}\left\{H_i\sum_{q=1}^{n-s}\left\{R_q\binom{n-s}{1}\left(\frac{1}{n-s}\right)\left(1-\frac{1}{n-s}\right)^{j-1}\right\}\right\}+\sum_{i=m'(m)-n+s+1}^{m'(m)-2}\left\{H_i\sum_{q=1}^{m'(m)-i}\left\{R_q\binom{m'(m)-i}{1}\left(\frac{1}{m'(m)-i}\right)\left(1-\frac{1}{m'(m)-i}\right)^{j-1}\right\}\right\}$$

$$\left.+H_{m'(m)-1}R_{m'(m)}\right] \tag{5.50}$$

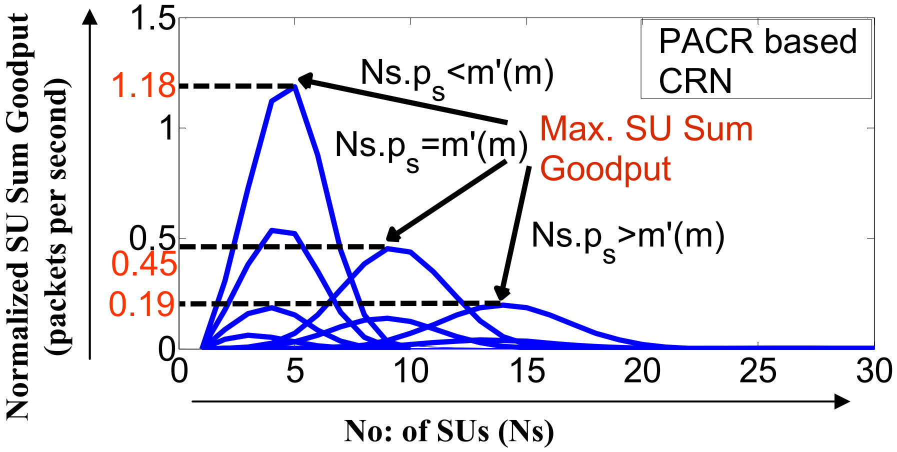
SU Sum Goodput as derived in (5.50) is normalized and plotted with *Ns* for different values of *m′(m)*. Three conditions are illustrated in Fig. 5.18 corresponding to the number of SUs being less than, equal to and greater than the number of idle channels. Mathematically, these conditions refer to $Ns.p_s<m'(m)$, $Ns.p_s=m'(m)$ and $Ns.p_s>m'(m)$.

Rise in PU activity is modeled in Fig. 5.18 by increasing values of p_r. Hence, *m′(m)* decreases, denoting a CRN with high traffic load. It is observed that under low PU activity, CRN_BASIC (same as CRN_UNRESERV) records the highest SU Sum Goodput.

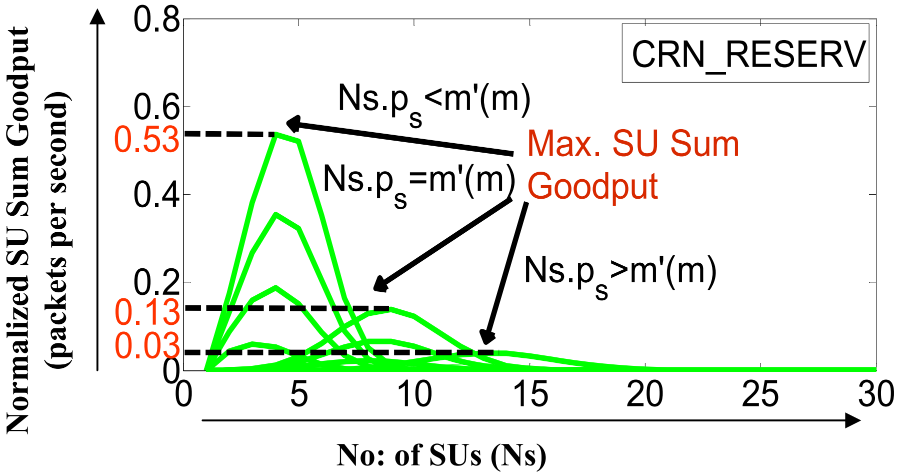
However, it also experiences maximum number of dropping instances, as indicated by more lines in Fig. 5.18 (a) compared to *PACR* based CRN in Fig. 5.18 (b) and CRN_RESERV in Fig. 5.18 (c). Moreover, SU Sum Goodput is higher in *PACR* based CRN than CRN_RESERV for all the three conditions as shown in Fig. 5.17 (b) and Fig. 5.17 (c).



(a)



(b)



(c)

Fig. 5.18 Variation in Normalized SU Sum Goodput with total number of SUs in (a) CRN_ BASIC (b) *PACR* based CRN and (c) CRN_RESERV for different number of idle channels

The normalized SU Sum Goodput is also plotted with different number of idle channels for *PACR* based CRN, CRN_BASIC and CRN_RESERV in Fig. 5.19. It is observed that while CRN_BASIC yields the maximum goodput, CRN_RESERV records minimum number of dropping instances. Comparing the product of SU Sum Goodput and number of instances for which SU remains in its channel without performing handoff, *PACR* based CRN provides the highest value of 0.33 as shown in Fig. 5.19.

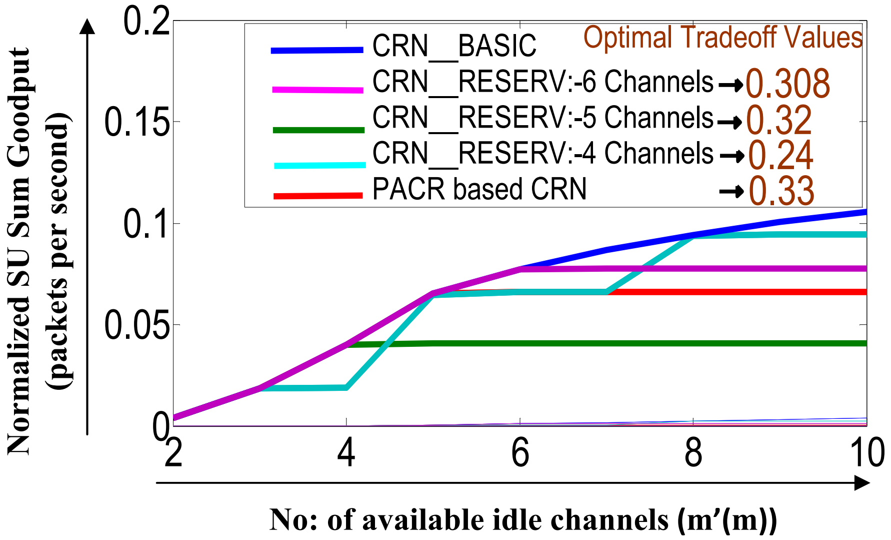


Fig. 5.19 Variation in Normalized SU Sum Goodput with no. of idle channels for CRN_BASIC, CRN_RESERV and *PACR* based CRN

Thus, *PACR* improves system performance, as it yields higher goodput than CRN_RESERV under low PU activity and records lower dropping instances than CRN_BASIC during high PU activity.

(ii) ***<u>Inferences from the Mathematical Models</u>***

Some significant inferences are drawn both from the system perspective and the user perspective based on this mathematical framework, which highlight the efficiency of *PACR* algorithm and are outlined as follows.

1. ***System Perspective: The overall system capacity in PACR based CRN increases by 's' compared to CRN_RESERV.***

Proof:

It is derived in [5.11] that the number of channels available for SUs in CRN_RESERV (denoted by N_{reserv}) is n, where $m'(m)-n$ number of channels are reserved for PUs.

Let RT and NRT users access zone 3 with probabilities P_{voice} and P_{data} respectively. Hence, the number of channels available to NRT and RT users as per *PACR* algorithm are given by,

$$N_{NRT} = (1 + P_{data})s \tag{5.51}$$

$$N_{RT} = n + s(P_{voice} - 1) \tag{5.52}$$

The total number channels for SU transmission is calculated as,

$$N_{PACR} = N_{NRT} + N_{RT} = n + s(P_{voice} + P_{data}) \tag{5.53}$$

Since, either RT or NRT user is sure to get access to an idle channel in zone 3, we have

$$P_{voice} + P_{data} = 1 \tag{5.54}$$

$$\Rightarrow N_{PACR} = n + s = N_{RESERV} + s \tag{5.55}$$

2. ***User Perspective: The acceptance probability in PACR based CRN is greater for RT users than NRT users under equally likely probabilities of their arrival.***

Proof:

The acceptance probabilities for RT and NRT users are given by,

P_{RT_acc}=(No. of idle channels in zone 4+Probability that there is no NRT user* No. of idle channels in zone 3)/{$m'(m)-n$}

$$= \frac{\left[\sum_{i=n+s}^{m'(m)}\left\{(1-Pr_i)\times\prod_{j=(n+s)-1}^{i-1}Pr_j\right\}\right]+\left[\sum_{i=n}^{n+s}\left\{(1-Pr_i)\times(1-P_{di})\times\prod_{j=n-1}^{i-1}Pr_j\right\}\right]}{m'(m)-n} \tag{5.56}$$

$$= \frac{x_1 + y_1}{m'(m) - n} \tag{5.57}$$

where x_1,y_1 are the two corresponding terms in the numerator of (5.56).

P_{NRT_acc}=(No. of idle channels in zone 2+Probability that there is no RT user* No. of idle channels in zone 3)/{2s}

$$= \frac{\left[\sum_{i=n-s}^{n}\left\{(1-Pr_i)\times\prod_{j=n-s-1}^{i-1}Pr_j\right\}\right]+\left[\sum_{i=n}^{n+s}\left\{(1-Pr_i)\times(1-P_{vi})\times\prod_{j=n-1}^{i-1}Pr_j\right\}\right]}{2s} \tag{5.58}$$

$$= \frac{x_2 + y_2}{2s} \tag{5.59}$$

where x_2, y_2 are the two corresponding terms in the numerator of (5.58). Considering equal intervals in each zone, $2s=m'(m)-n$. Hence from (5.57) and (5.59),

$$P_{RT\_acc} - P_{NRT\_acc} = \frac{(x_1 + y_1 - x_2 - y_2)}{\{m'(m) - n\}} \tag{5.60}$$

For equally likely arrival probability of RT and NRT users,

$$P_{voice} = P_{data} = 0.5 \Rightarrow y_1 - y_2 = 0 \tag{5.61}$$

Therefore, from (5.60),

$$P_{RT\_acc} - P_{NRT\_acc} = \frac{(x_1 - x_2)}{\{m'(m) - n\}} \tag{5.62}$$

Let us consider x_1-x_2. It is seen that both the summation intervals are in the interval of n and hence, the same. Further,

$$Pr_i = 1 \Rightarrow Pr_j = 1 \;\forall\; j = 1 \;\; to\; i-1 \tag{5.63}$$

Hence, as k and i increase, $\prod_{j=k}^{i} Pr_j$ decreases. Thus from (5.56), (5.58) and (5.60), we have

$$x_1 - x_2 > 0 \Rightarrow P_{RT\_acc} > P_{NRT\_acc} \tag{5.64}$$

3. ***System Perspective: The Secondary User blocking probability decreases from zone 2 to zone 4.***

Proof:

Let Pr_j be the probability of PU arrival in jth channel. Let P_{SU_zonei} denotes the probability that SU is allowed to transmit in any of the channels in zone i. P_{b_zonei} provides the corresponding blocking probability for that SU in zone i. As per *PACR* algorithm, in zone 1,

$$Pr_j = 1 \tag{5.65}$$

$$\Rightarrow P_{SU\_zone1} = 0 \Rightarrow P_{b\_zone1} = 1 - P_{SU\_zone1} = 1 \tag{5.66}$$

In zone 2,

$$P_{SU\_zone2} = \frac{\left[\sum_{i=n-s}^{n} \left\{ (1 - Pr_i) \times \prod_{j=n-s-1}^{i-1} Pr_j \right\} \right]}{s} \tag{5.67}$$

$$\Rightarrow P_{b\_zone2} = P_{b\_zone1} \times \left(1 - P_{SU\_zone2}\right) = 1 - P_{SU\_zone2} \tag{5.68}$$

In zone 3,

$$P_{SU\_zone3} = \frac{\left[\sum_{i=n}^{n+s} \left\{ (1 - Pr_i) \times \prod_{j=n-1}^{i-1} Pr_j \right\} \right]}{s} \tag{5.69}$$

$$\begin{aligned} \Rightarrow P_{b\_zone3} &= P_{b\_zone2} \times \left(1 - P_{SU\_zone3}\right) \\ &= \left(1 - P_{SU\_zone2}\right)\left(1 - P_{SU\_zone3}\right) \end{aligned} \tag{5.70}$$

In zone 4,

$$P_{SU\_zone4} = \frac{\left[\sum_{i=n+s}^{m'(m)} \left\{ (1 - Pr_i) \times \prod_{j=(n+s)-1}^{i-1} Pr_j \right\}\right]}{m'(m) - (n+s)} \tag{5.71}$$

$$\Rightarrow P_{b\_zone4} = P_{b\_zone3} \times \left(1 - P_{SU\_zone4}\right)$$

$$= \left(1 - P_{SU\_zone2}\right)\left(1 - P_{SU\_zone3}\right)\left(1 - P_{SU\_zone4}\right) \tag{5.72}$$

It is obvious from (5.66), (5.68), (5.70) and (5.72) that

$$P_{b\_zone4} < P_{b\_zone3} < P_{b\_zone2} < P_{b\_zone1} \tag{5.73}$$

4. ***System Perspective: The optimal value of s is equal to n/2 for low PU activity and (m'(m)-n)/2 for high PU activity.***

Proof:

As per *PACR* algorithm, in times of low PU activity, the total number of reserved channels for PU is decreased by *s*. Let us consider the region from 1 to {*m'(m)-n*} where *m'(m)-n=n*. Hence, (*n-s*) channels are reserved for PU. The number of idle channels occupied by SU is given by,

$$m_{SU} = s \times p_s \tag{5.74}$$

where p_s = probability of idle channel access by SU. The number of idle channels that PU can access without affecting the existing SU transmissions is expressed as,

$$m_{PU} = n - m_{SU} = n - s \times p_s \tag{5.75}$$

The higher the value of m_{PU}, the more the SU is free from PU interference. Thus, m_{PU} is a measure of uninterrupted SU transmission. Hence, the product of SU channel utilization and uninterrupted transmission is given by,

$$Prod = m_{SU} \times m_{PU} = s \times p_s \times \left(n - s \times p_s\right) \tag{5.76}$$

The objective is to find the optimal value of s such that the value of *Prod* is maximum. Differentiating *Prod* with respect to s and equating it to 0,

$$\frac{dProd}{ds}=\frac{d\{s\times p_s\times(n-s\times p_s)\}}{ds}=0\Rightarrow s=\frac{n}{2p_s} \tag{5.77}$$

Taking the second order derivate of *Prod* with respect to n,

$$\frac{d^2Prod}{ds^2}=-2p_s^2<0 \tag{5.78}$$

Hence, Prod has the maximum value when $s=\dfrac{n}{2p_s}$. (5.79)

Considering $p_s = 1$ for all SUs (denoting channel access by SU with absolute certainty) (5.79) is modified as $s = \dfrac{n}{2}$ (5.80)

Therefore, the optimal value of s is $n/2$ for low PU activity.

Again, let us consider the region from n to $m'(m)$. In times of high PU activity, the number of channels reserved for PU is increased by s from the default value. Hence, proceeding in a similar way,

$$m_{SU}=\{m'(m)-(n+s)\}\times p_s \tag{5.81}$$

$$m_{PU}=\{m'(m)-n\}-\{m'(m)-(n+s)\}\times p_s \tag{5.82}$$

Let $x=m'(m)-(n+s)$, $y=m'(m)-n$. Taking the product of (5.81) and (5.82) and differentiating *Prod* with respect to x and equating it to 0,

$$\frac{dProd}{dx}=\frac{d\{x\times p_s\times(y-x\times p_s)\}}{dx}=0\Rightarrow x=\frac{y}{2p_s} \tag{5.83}$$

Also, $$\frac{d^2Prod}{dx^2}=-2p_s^2<0 \tag{5.84}$$

Considering $p_s = 1$ for all SUs (denoting channel access by SU with absolute certainty) (5.83) is modified as

$$x=\frac{y}{2}\Rightarrow s=\{m'(m)-n\}/2 \tag{5.85}$$

Hence, the optimal value of s is $\{m'(m)-n\}/2$ for high PU activity.

5. ***User Perspective: QoS for VoIP is maintained by RT users as the dropping probability is the least for them.***

Proof:

As VoIP communication occurs in the form of talkspurts [5.17], the VoIP traffic for a single SU is formulated as an on-off model with $1/\alpha$ and $1/\beta$ as on and off periods, respectively, that are exponentially distributed [5.21]. Therefore, the probability with which SU is busy transmitting is given by,

$$P_{busy} = \frac{\alpha^{-1}}{\alpha^{-1} + \beta^{-1}} \tag{5.86}$$

For n idle channels, VoIP throughput is given by,

$$\begin{aligned} C_{VoIP} = P_{busy}\, R_v \Big\{ & (1 - Pr_1) + Pr_1 (1 - Pr_2) + \\ & Pr_1 Pr_2 (1 - Pr_3) + \ldots + \prod_{i=1}^{n-1} Pr_i (1 - Pr_n) \Big\} \end{aligned} \tag{5.87}$$

where R_v is the VoIP transmission rate. The various terms in (5.87) correspond to the spectrum handoff instances for RT users with increase in PU activity. Finally, after the nth term, RT user is dropped from CRN. Hence, with rise in the probability of reaching to the nth term in (5.87), the dropping probability also increases for RT user. When $m'(m)-n$ number of channels is reserved for PU, the probabilities of occupying a reserved channel and an unreserved channel by PU in CRN comprising of Np number of PUs are given by (5.88) and (5.89) respectively.

$$Pr_{reserv} = \binom{m'(m)-n}{1} \left(\frac{1}{m'(m)-n} \right) \left(1 - \frac{1}{m'(m)-n} \right)^{Np-1} \tag{5.88}$$

$$Pr_{unreserv} = \prod_{i=1}^{m'(m)-n} Pr_{reserv} \binom{n}{1} \left(\frac{1}{n} \right) \left(1 - \frac{1}{n} \right)^{Np-1} \tag{5.89}$$

As more number of channels is reserved for PUs, $Pr_{unreserv}$ decreases due to the multiplication term in (5.89). Therefore, when the maximum number of channels is reserved for PU, $Pr_{unreserv}$ is the lowest. This actually happens when $m'(m)-n+s$ number of channels are reserved for PU, while RT SUs still transmit in zone 4. As $Pr_{unreserv}$ decreases, the probability of spectrum handoff in zone 4 also decreases. This reduces the probability of reaching to the nth term in (5.87), thereby decreasing the overall dropping probability for RT users.

5.7 Performance Evaluation and Discussion

This section performs detailed analysis of the advantages of the channel reservation policy in the CRN with respect to the individual SUs and the overall system, respectively. Initially, the analytical models designed in Section 5.4 are used to study the VoIP call quality for each SU. In addition, the simulation models for the VoIP based CRN are designed in OPNET Modeler 16.0.A. to analyze and validate the performance gain in CRN_RESERV with respect to CRN_UNRESERV as evaluated in Section 5.4. Finally, the simulation model is enhanced further to incorporate the *PACR* Algorithm. Subsequently, the performance gain of *PACR* based CRN is established and its mathematical framework is validated.

5.7.1 Performance Study for VoIP SUs in CRN_RESERV – User Perspective

The designed mathematical models for the VoIP based CRN are analyzed in MATLAB with the modeling of VoIP codecs and the formulation of VoIP traffic in the CRN comprising of multiple PUs and SUs. The quality metrics under evaluation include the successful transmission time for a VoIP call, utilization of the total on time in a transmission interval, the spectrum handoff delay, the total number of successfully delivered packets and the corresponding packet loss, and finally the VoIP call quality in terms of MOS.

Initially, different VoIP talkspurt characteristics are applied to the SU traffic based on the silence detectors used in modern communication [5.25] as described in Table 5.2.

Table 5.2 VoIP Traffic Parameters for Different Silence Detectors

Sl. No.	Silence Detector	Mean Spurt in ms	Mean Gap in ms	Talkspurt percentage
A	P.59 without hangover	227	596	27.6%
B	P.59 with hangover	1004	1587	38.7%
C	Brady with hangover	1200	1800	40%
D	Sriram without hangover	352	650	35.1%
E	G.729B (dynamic hangover)	362	488	42.6%
F	NeVoT SD (short hangover)	326	442	42.5%
G	NeVoT SD (default hangover)	903	1216	42.6%

The corresponding successful transmission time for a single VoIP SU in a particular channel prior to spectrum handoff is plotted in Fig. 5.20 that shows significant increase in CRN_RESERV compared to CRN_UNRESERV. This is because channel reservation ensures lower handoff delay and minimizes the interference with respect to a SU. With the decrease in the probability of spectrum handoff for the SU in CRN_RESERV, the corresponding utilization of the total transmission time interval t_d by that SU prior to any PU arrival increases compared to CRN_UNRESERV as seen from Fig. 5.21.

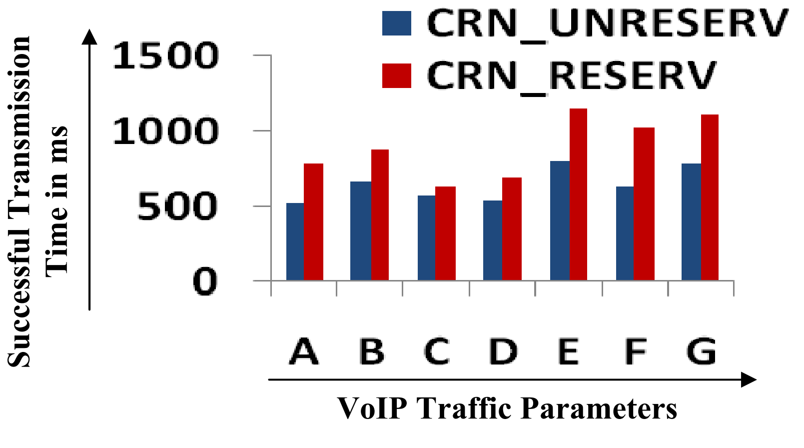


Fig. 5.20 Effect of VoIP traffic parameters on the Successful Transmission Time by a single SU in CRN_RESERV and CRN_UNRESERV

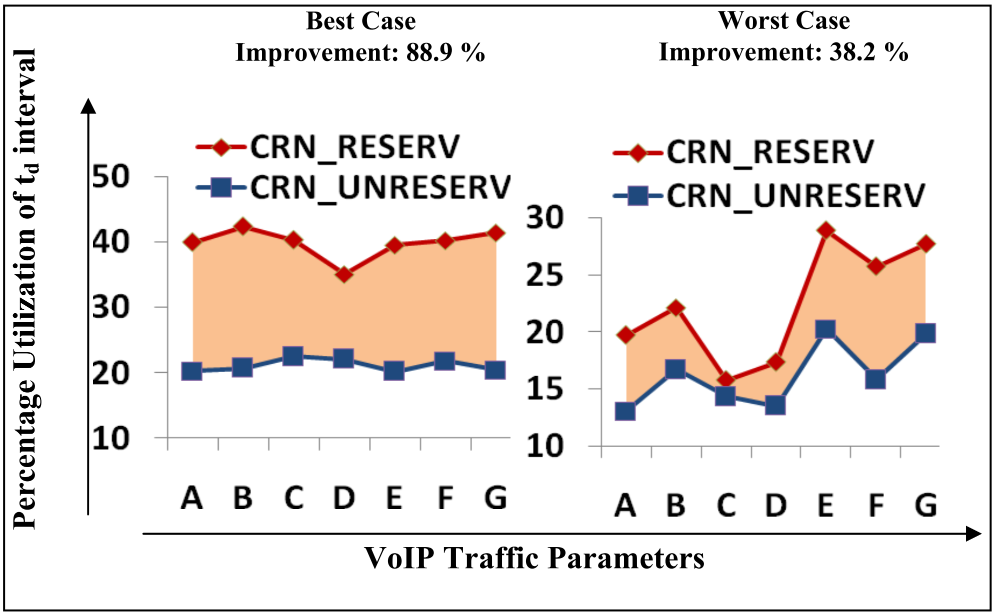


Fig. 5.21 Percentage Utilization of the total transmission time t_d by a single SU in CRN_RESERV and CRN_UNRESERV in the best and worst case with respect to PU activity under different VoIP parameters

The best and the worst case scenarios with respect to the PU activity in Fig. 5.21 record a mean improvement of 89% and 38%, respectively for CRN_RESERV over CRN_UNRESERV. This increase in the utilization, as illustrated by the shaded region in Fig. 5.21 is a measure of the extent to which a SU remains independent from any PU activity. It is observed from Fig. 5.21 that CRN_RESERV is more effective for the VoIP conversations having higher talkspurt percentage and lesser number of talkspurts.

In order to study the effect of VoIP traffic characteristics on the spectrum handoff delay, different number of talkpsurts is considered. Based on the average value of the talkspurt percentage as observed from Table II, the overall talkspurt percentage is kept at a constant value of 40%. It is seen from Fig. 5.22 that reserving the channels for the PUs decreases the delay incurred during the spectrum handoff by 95% in CRN_RESERV, which is within tolerable limits for the VoIP applications. CRN_UNRESERV, on the other hand, experiences a high amount of delay that is completely unacceptable for the real-time communication.

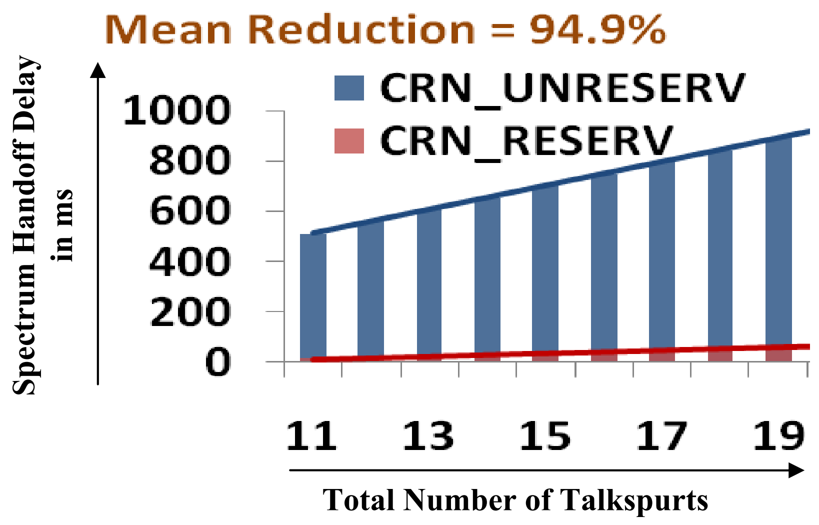


Fig. 5.22 Variation in the Spectrum Handoff Delay with different number of talkspurts in CRN_RESERV and CRN_UNRESERV

Moreover, as the number of talkspurts increases, the VoIP transmission occurs in short cycles of on time and off time. This increases the probability of the PU arrival during the on time (t_{on}), thereby resulting in higher delay due to spectrum handoff for both CRN_RESERV and CRN_UNRESERV.

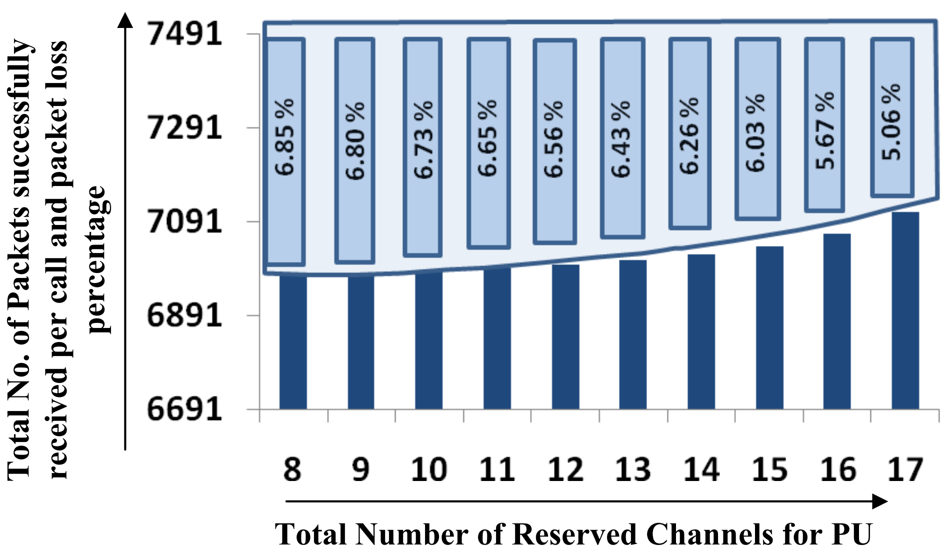


Fig. 5.23 Effect of different number of channels reserved for the PUs on the SU Throughput and the corresponding Packet Loss Percentage

Apart from the nature of the VoIP traffic, the total number of channels reserved for the PUs also plays a critical role in determining the success of the VoIP calls. Let CRN_RESERV comprises of 20 channels. From the network perspective, the optimal trade-off between an increase in throughout and a higher interference level is provided by reserving the suitable number of channels as given by (5.19). However, the individual VoIP SUs perform better on increasing the number of reserved channels due to a drastic reduction in the interference with the PUs. This results in a higher number of packets being successfully delivered as shown in Fig. 5.23. The corresponding packet loss also decreases and remains within the threshold level for proper VoIP communication.

The combined effect of the VoIP traffic characteristics and the number of channels reserved is studied with respect to a successful VoIP transmission. Fig. 5.24 is the plot for the percentage utilization of a single transmission slot (t_{on}) by a VoIP SU for different number of talkspurts and reserved channels. A higher utilization of t_{on} is recorded with the rise in the number of reserved channels and at lower number of talkspurts due to the decrease in the spectrum handoff delay and the total time of interference.

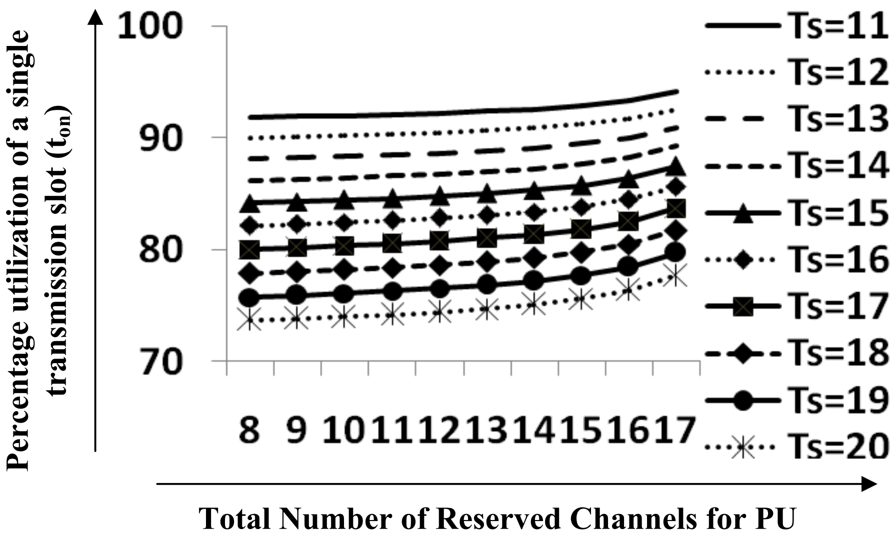


Fig. 5.24 Percentage Utilization of a single transmission slot (t_{on}) by a VoIP SU for different talkspurt values and varying number of reserved channels

Finally, the VoIP call quality, as determined by the Mean Opinion Score (MOS), is calculated for diverse channel reservation scenarios with respect to different VoIP traffic characteristics. The Mean Opinion Score gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using various codecs [5.23]. The E-model as specified by ITU-T G.107 is used to calculate the MOS based on the following equation.

$$MOS = 1, \forall R \leq 0$$
$$= 1 + 0.035R + R(R-60)(100-R)7 \times 10^{-6}, \forall 0 < R < 100$$
$$= 4.5, \forall R \geq 100 \quad where \quad R = R_O - I_s - I_d - I_{e,eff} + A \tag{5.90}$$

R_o is the basic signal-to-noise ratio. I_s, I_d and I_e represent all the impairments that occur with the voice signal, delay and echo effects and low bit-rate codecs, respectively. The term A is an *advantage factor*, which represents the advantage of access for certain systems relative to the conventional systems, trading the voice quality for convenience.

It is observed from Fig. 5.25 that the MOS value increases with a rise in the talkspurt percentage and remains above the threshold value of 3 in all the scenarios. Moreover, the drop in the contour lines with a rise in the number of reserved channels indicates an increase in the MOS value for the ongoing VoIP calls.

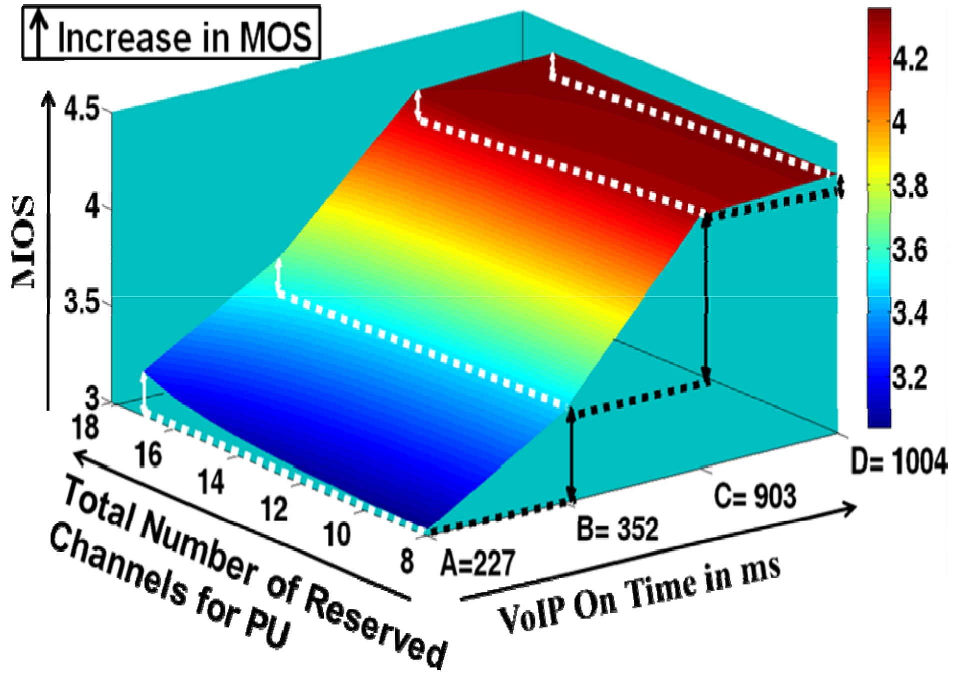


Fig. 5.25 Improvement in the VoIP Call Quality in terms of MOS value with an increase in the number of reserved channels and a rise in the talkspurt percentage

Thus, it is finally inferred that *a higher talkspurt percentage with lower number of talkspurts coupled with an increase in the number of reserved channels for the PUs results in a significant increase in the overall quality of VoIP transmission and makes CRN_RESERV capable of hosting the real-time VoIP applications*.

5.7.2 Performance Analysis of the VoIP SUs in the Simulation Model – System Perspective

The second phase of performance study is done in the simulation models to evaluate the performance gain of CRN_RESERV from the overall system perspective. In order to observe the aspects of real-life like communication under channel reservation policy, CRN_RESERV is implemented in the simulation model for the VoIP based CRN, that has already been designed in the OPNET Modeler 16.0.A. in Chapter 2. Fig. 5.26 is a screenshot of an actual scenario for CRN_RESERV with respect to two channels that are utilized by the SUs to transmit. It is observed that enabling channel reservation delays the arrival of the PUs in those channels. Thus, the SUs continue uninterrupted transmission for a longer duration of time before being dropped from the channels. The corresponding bursty traffic of the VoIP applications is clearly identified in the figure.

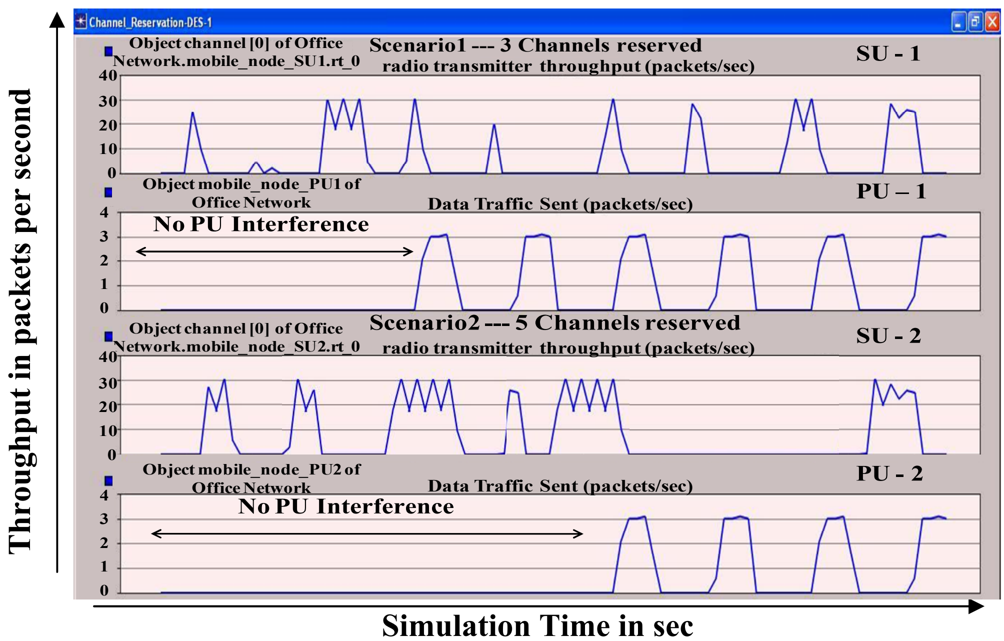


Fig. 5.26 Screenshot of an ongoing simulation in the OPNET Modeler 16.0.A. depicting channel access by the SUs for different number of reserved channels

As channel reservation decreases the spectrum utilization by the SUs, it is observed from Fig. 5.27 that CRN_RESERV records lower SU throughput than CRN_UNRESERV. With the decrease in the number of reserved channels for the PUs, more number of SUs is admitted in the CRN, thereby leading to a rise in the total system throughput as illustrated in Fig. 5.27. However, this results in an increase in the interference level and a rise in the number of invalid packets received by the SUs. Fig. 5.28 illustrates the time average plot of the fraction of time for which the invalid (or erroneous) packets are received by the receivers at the SUs. It is seen that CRN_UNRESERV produces the invalid packets for 4/5th of the total transmission time, that is significantly reduced in CRN_RESERV. The total number of dropping instances for the SUs also decreases considerably in CRN_RESERV as shown in Fig. 5.29.

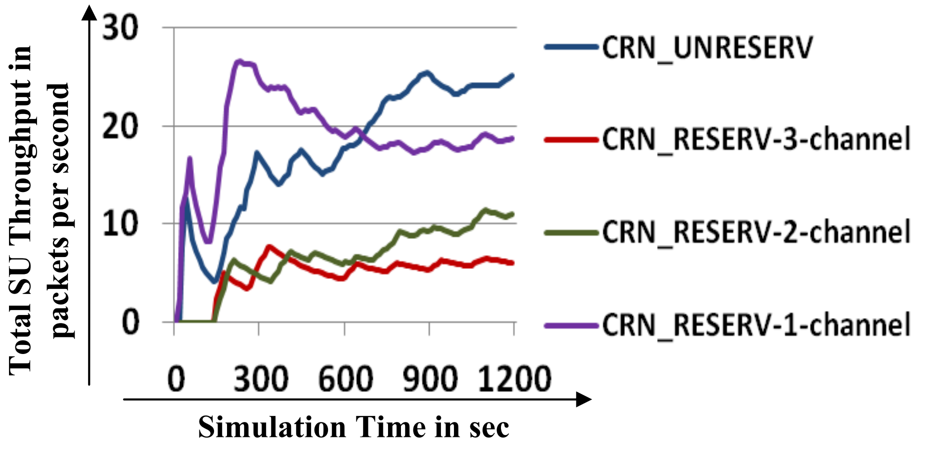


Fig. 5.27 Variation in the total SU Throughput (Time Average) for different CRN Scenarios

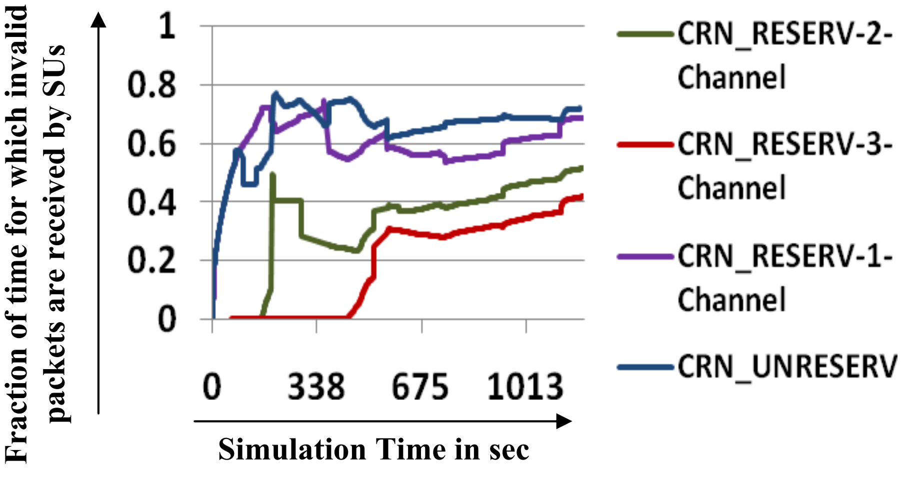


Fig. 5.28 Fraction of time (Time Average) for which the invalid packets are received by a SU under different CRN Scenarios

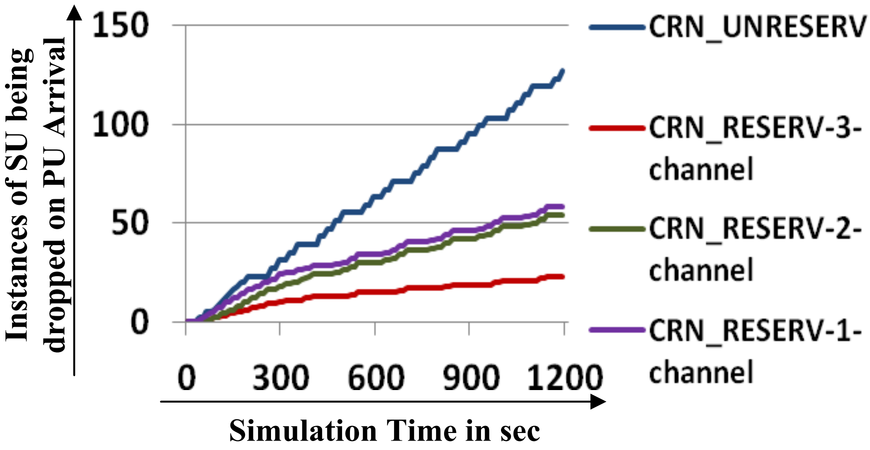


Fig. 5.29 Instances (Sample Sum) of the SU being dropped on the arrival of a PU with respect to different CRN scenarios

The performance improvement of CRN_RESERV over CRN_UNRESERV is summarized in Table 5.3. It is clearly observed that the optimal tradeoff is achieved by reserving 2 channels for the PUs in the CRN consisting of 4 channels, thereby satisfying the condition in (5.19).

Table 5.3 Performance Evaluation of CRN_RESERV with respect to CRN_UNRESERV

Network Parameters	Performance Statistics	Number of Channels reserved		
		1-channel	2-channel	3-channel
System Throughput	% Decrease	25.71%	55.98%	76.04%
Dropping Instance	% Decrease	54.33 %	57.48%	81.88%
Erroneous Packets received	% Decrease	4.67%	28.53%	41.81%

5.7.3 Performance Evaluation of *PACR* Algorithm

In the final phase, the developed CRN_RESERV in OPNET Modeler 16.0.A. is further modified to incorporate the components of the *PACR* Algorithm namely, *ACR* and *PBA* in the suitable process models belonging to MAC, PHY and Application layers.

Initially, the total traffic received is plotted with respect to simulation time for CRN_BASIC, CRN_RESERV and *PACR* based CRN in Fig. 5.30. It records over 100% increase in system throughput for *PACR* based CRN over CRN_RESERV and thus, validates the inference drawn from mathematical model in Section 5.6.

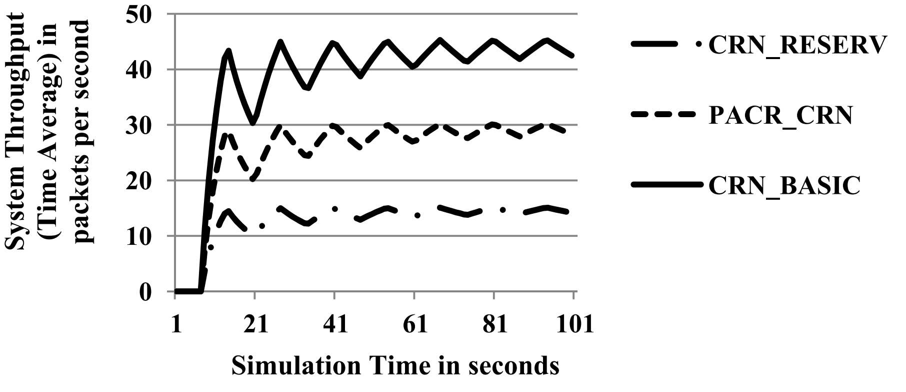


Fig. 5.30 Time Average System Throughput for CRN_BASIC, CRN_RESERV and *PACR* based CRN

This outcome is also confirmed by the time average transmission counts for the three systems in Fig. 5.31. It is imperative that the higher the dip in transmit count, the more the number of dropping instances. As seen from the figure, application of *PACR* records lesser number of dropping instances compared to CRN_BASIC and thus confirms the inference drawn in Section 5.6.

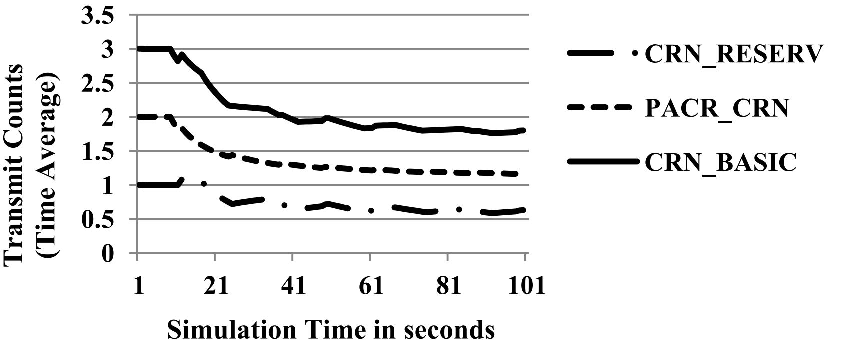


Fig. 5.31 Time Average Transmission Counts for CRN_BASIC, CRN_RESERV and *PACR* based CRN

It has further been proved in Section 5.6 that RT users have more acceptance probability in *PACR* based CRN over NRT users. This is confirmed in Fig. 5.32 which illustrates an increase in throughput for RT users compared to NRT users. In addition, the fact that RT users witness the lowest dropping probability is validated in Fig. 5.33. It is seen that while both RT and NRT users suffer decrease in transmission counts with increase in PU activity, the dip is higher in NRT users resembling higher dropping instances as compared to RT users.

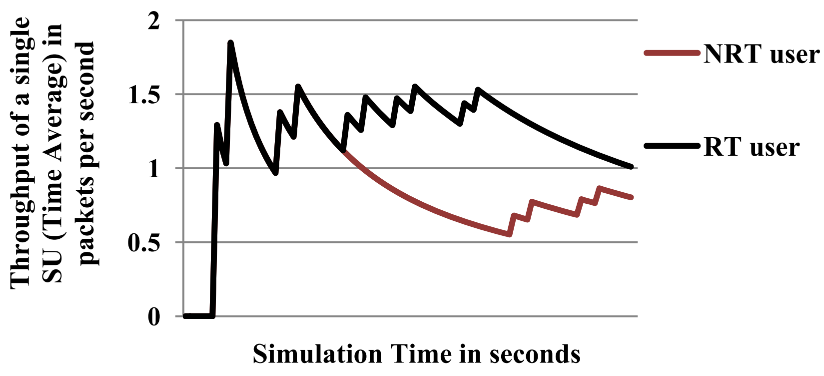


Fig. 5.32 Time Average Throughput for a single RT and NRT user in *PACR* based CRN

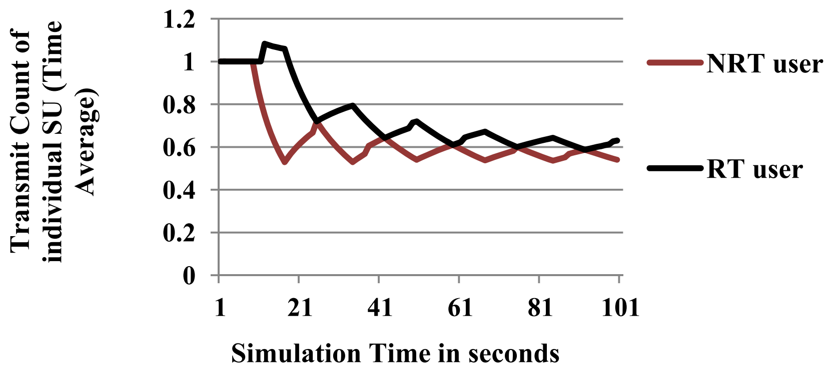


Fig. 5.33 Time Average Transmit Count for a single RT user and NRT user in *PACR* based CRN

Finally, the screenshot of the ongoing simulation in Fig. 5.34 demonstrates the behavior of RT and NRT users with PU arrival. While NRT user has to vacate the channel repeatedly, leading to increased dropping counts, RT user enjoys longer periods of successful transmission without PU interference.

Overall in this section, it can be inferred that the analytical model for the VoIP based CRN proves the necessity of the channel reservation scheme from the user perspective for maintaining the QoS in real-time applications. The simulation model, on the other hand, points out the advantages of CRN_RESERV as a system and further validates the inference drawn in Section 5 that an optimal tradeoff must be maintained between system utilization and the SU dropping probability through careful selection of the number of reserved channels for the PUs. Finally, the advantages of adaptive channel reservation through the dersigned *PACR* Algorithm are established in simulation models.

Thus, the contribution of this work in terms of a complete performance evaluation as stated in Table 5.1 is successfully established.

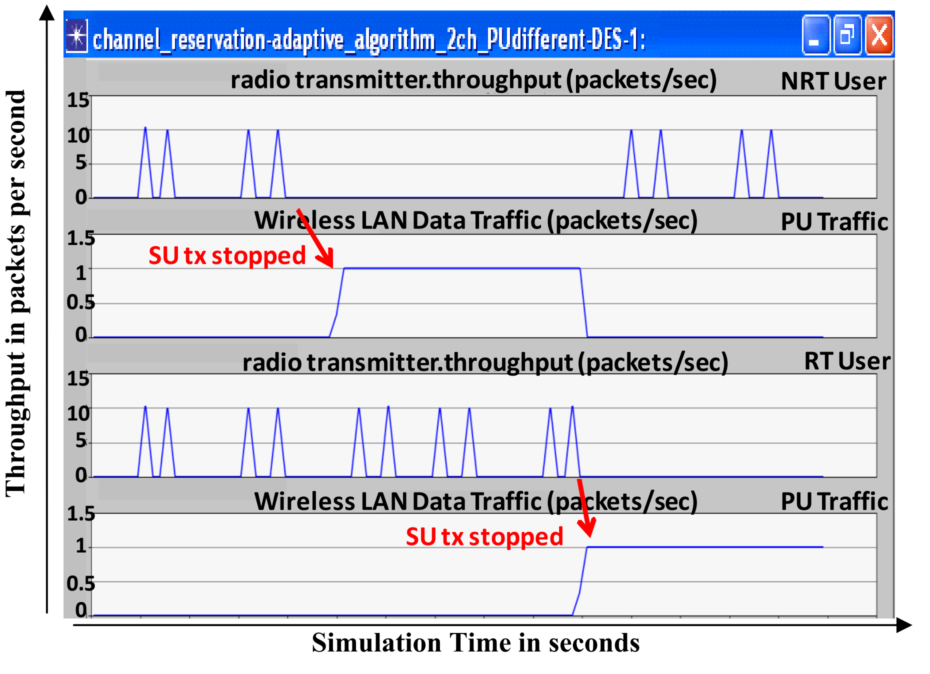


Fig. 5.34 Screenshot of ongoing simulation in OPNET Modeler 16.0.A. depicting channel access by RT and NRT SUs in *PACR* based CRN

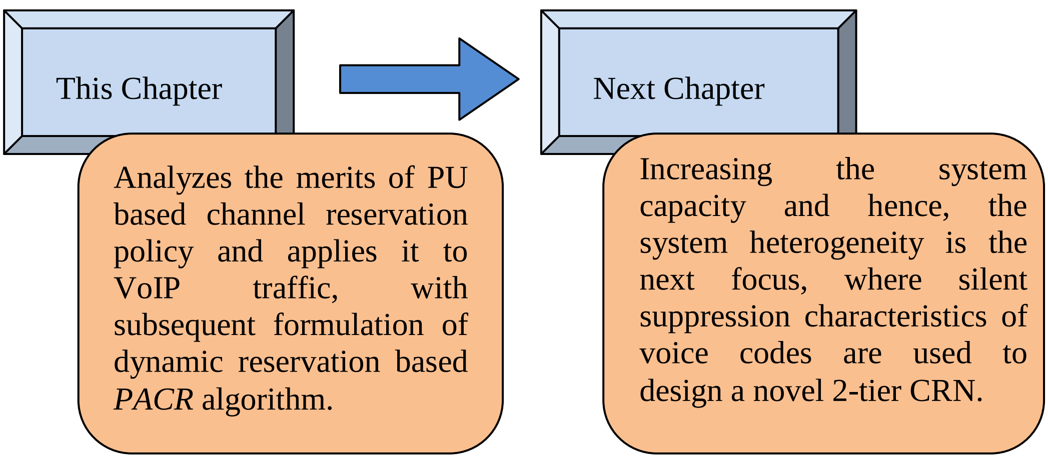
5.8 Summary

This chapter has carried out a detailed performance evaluation of the PU based channel reservation policy in CRN. The developed architectural framework highlights the need for coordination among the MAC protocols, channel assignment schemes, spectrum handoff policies and spectrum broker functionalities for the practical implementation of the channel reservation strategy. Thereafter, the analytical models are designed to determine the optimal tradeoff between the decrease in throughput and a rise in the handoff instances for the SUs by reserving suitable number of channels for the PUs. Comparative performance evaluation in the simulation models reflects a decrease in the dropping, handoff and blocking probabilities with over 50% increase in the instances of interference-free transmission for the SUs.

In addition, the channel reservation policy is applied to provide adequate QoS to the VoIP applications in the CRN. Rigorous stochastic analysis provides the mathematical expressions for important SU performance parameters including the total time of interference, the spectrum handoff delay and the probabilities of accessing the reserved and unreserved channels under the conditions of both high and low PU activity. A significant reduction is observed in the spectrum handoff delay and the packet loss percentage for the VoIP calls with the mean MOS value of greater than 3, thereby confirming the high quality of VoIP transmission for the SUs.

Thereafter, this chapter has addressed the problem of static channel reservation scheme for PU in CRN by introducing a novel *PACR* algorithm that adaptively reserves channels based on PU activity and performs priority based channel allocation to RT and NRT users. Stochastic analysis in the mathematical models reflects performance improvement in CRN with increased system capacity in terms of SU Sum Goodput. System heterogeneity is further increased as *PACR* enables CRN to successfully host both VoIP and data applications. Simulation results confirm an additional 100% improvement in throughput for *PACR* based CRN with effective allocation of idle channels to RT and NRT users, compared to static channel reservation policy. QoS is also ensured for the VoIP applications in RT users in terms of their lowest handoff and dropping probabilities and higher counts of successful transmission than NRT users.

The outcome of this study has been published in ***Computers and Electrical Engineering Journal'15, Elsevier (SCI Indexed)*** and also in the ***International Conference Proceedings of IEEE ICACCI'13 (Mysore) and IEEE CSE'14 (China)***.



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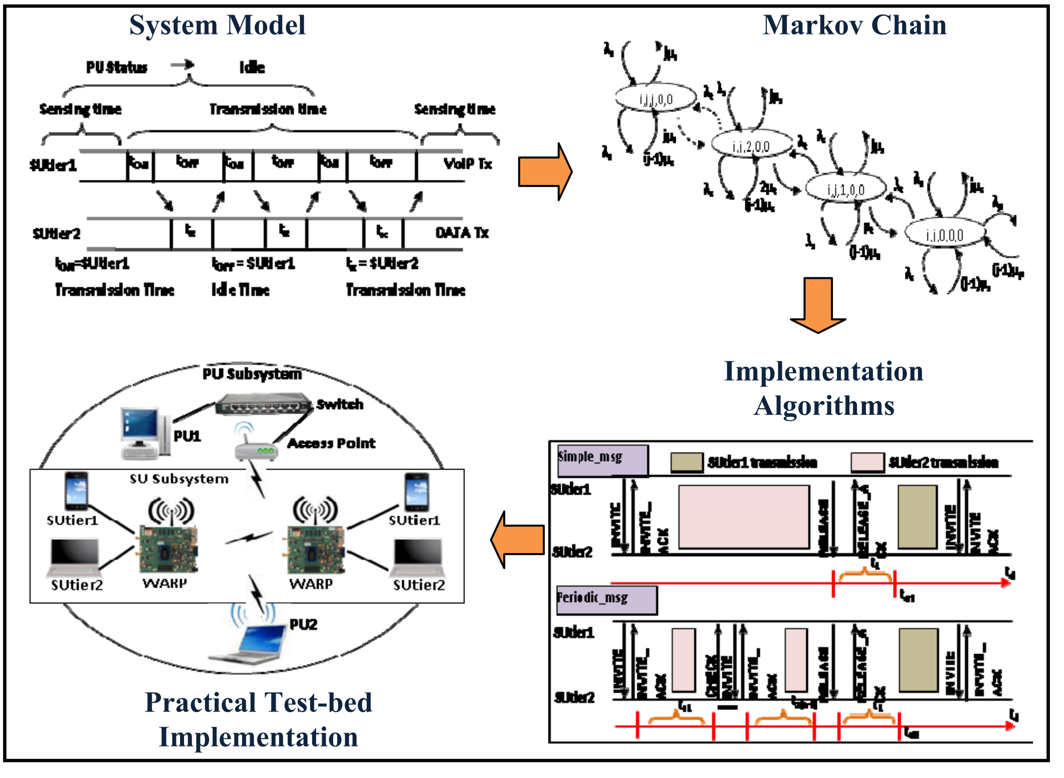
Chapter 6.

VOIP BASED 2-TIER CRN FOR IMPROVED SPECTRUM UTILIZATION

Chapter Highlights



CHAPTER 6: VoIP based 2-Tier CRN for Improved Spectrum Utilization

"We are living in the fourth great network revolution – the marriage of computing and connectivity. Some have labeled it the information age."

- Tom Wheeler, FCC Chairman

Outline of the Chapter

While previous chapters have focused on securing adequate QoS for successful VoIP communication by the SUs, increasing overall spectrum utilization is one crucial aspect of CRN that has not yet been thoroughly explored. It is quite imperative that there has to be a trade-off between the total number of admissible users in CRN and the QoS to be provided to their applications. As the goal of overlay based CRN [6.1] is to maximize spectrum utilization by allowing SUs to access spectrum bands in absence of the PUs, research focus is on enhancing capacity [6.2, 6.3] in CRN through strategies like spectrum handoff [6.4], channel reservation [6.5], Medium Access Control (MAC) protocols [6.6], etc. Also, energy efficiency has attracted much research recently as in [6.7]. However, the total system capacity in CRN is limited due to several factors such as SU and PU traffic arrival rates, transmission probabilities, handoff issues, etc [6.8].

In this regard, fixed upper bound on the maximum system capacity of CRN as derived in contemporary research studies [6.5], [6.8] limits further progress in research in this domain. It is, thus, implied that the existing policies in Basic CRN fail to record much increase in performance efficiency once and after CRN reaches its maximum capacity limit. ***This chapter aims to solve this problem by introducing the concept of '2-tier CRN' that not only increases overall system capacity but also allows the system to host diverse applications, thereby initiating a new direction of future research in this discipline.***

This chapter provides the novel design approach for VoIP based 2-tier CRN with every detail of formulation of the framework. The complex interaction among the primary and secondary users in the proposed system is studied extensively in mathematical, simulation and Markov models. Performance evaluation confirms 300% improvement in link utilization and hence, higher network throughput in 2-tier CRN as compared to Basic 1-tier CRN. From the practical implementation perspective, two algorithms namely, *Simple_msg* and *Periodic_msg* are further designed and the pros and cons of each of these techniques are discussed in analytical and OPNET based simulation models. Finally, the practical applicability of the designed network is established by successfully implementing it in a test-bed based disaster management system.

6.1 Introduction

VoIP based 2-tier CRN consists of one set of PUs and two sets of SUs. The first layer of SUs (SUtier1) performs VoIP communication during idle periods of PUs. Considering 40-50 percent of average talkspurt [6.9] for VoIP users, codecs implement silence suppression mechanisms to preserve bandwidth during silence periods of SUtier1. The second layer of SUs (SUtier2) performs non real-time data transmission in silence periods of SUtier1. On arrival of a new talkspurt or end of SUtier1 transmission time interval, SUtier2 stops transmission and transfers access of channel to SUtier1.

The motivation behind this work is that CRN is designed mainly for wireless communication [6.10], with voice communication as the main application. If PU traffic is modeled with on-off periods of activity [6.11], SUs with voice traffic can also be considered to follow on-off activity as voice traffic has intermittent idle time periods [6.12]. The idea is to exploit these idle time slots and admit a new tier of SUs in CRN. This section therefore carries out a detailed literature survey concerning the relevant aspects of VoIP and CRN domain, that lay the foundation towards the proposed design and implementation of two-tier CRN.

6.1.1 Literature Survey

The proposed model for VoIP based 2-tier CRN in this chapter comprises of three significant aspects, namely system capacity, system heterogeneity and application-oriented study with respect to VoIP. Accordingly, this section provides an outlook of the ongoing research in these domains and subsequently, highlights the novelty of the work in this chapter.

(i) Literature Survey: System Capacity in CRN

Current works have focused on increasing system capacity in CRN using different mechanisms. Firstly, call admission control is designed using slot-ALOHA in [6.13] and proactive admission control scheme in [6.14] to increase system capacity and reliability. Secondly, power allocation schemes are used to increase SU ergodic capacity in [6.15] and in absence of Channel State Information (CSI) in [6.16], along with the design of receiver and frame structure [6.3], joint subcarrier and power allocation policy [6.17] and using Karush–Kuhn–Tucker (KKT) scheme in [6.18]. Finally, Dynamic Spectrum Access (DSA) strategies are developed using predetermined access control mechanism [6.19], Game Theory [6.20], Renewal Theory [6.21] and myopic sensing and access policies [6.22]. Capacity scaling has also been performed on emerging networks by designing greedy network algorithm based Cognitive Wireless Sensor Networks in [6.23], MIMO empowered CRN in [6.24] and Cognitive Hybrid Division Duplex comprising of FDD on macrocell and TDD on cognitive femtocell in [6.25]. Increasing system capacity of Basic CRN in these works refer to either or both of two aspects namely, i) maximizing SU throughput, and ii) admitting new SUs in CRN by dropping other SUs. However, user capacity is limited by a fixed upper bound due to several factors (PU and SU traffic, arrival and transmission probabilities, mobility features, imperfect spectrum sensing effects, no. of idle channels and their behavior with time, etc.) as derived in [6.8]. This chapter focuses on further increasing user capacity by *admitting more SUs in CRN without dropping other SUs and allowing concurrent transmission of two SUs on the same channel* and, thus, increases the total system throughput.

(ii) Literature Survey: System Heterogeneity in CRN

Heterogeneity in CRN has been addressed with respect to i) varying channel characteristics by performing auction based spectrum sharing in [6.26], ii) integrating different networks such as design of Cognitive SMANET comprising of MANET and Sensor Networks in [6.27] and iii) heterogeneous SUs with their disparities in PHY/MAC systems as discussed in [6.28]. Specifically addressing the third category, [6.29] has classified users as high priority (Real Time or RT) and low priority (Non-Real Time or NRT) users and proposed DSA schemes. QoS management for RT users is studied by designing priority queues in [6.30], virtual queuing interface in [6.31] and spectrum allocation framework in [6.32]. Capacity of NRT users is also enhanced with sub-channel assignment and power distribution algorithms by designing resource allocation strategies in [6.33] and implementing "fast barrier" and "approximate" method in [6.34]. It is noticed that user heterogeneity in these works is supported by allowing prioritized access of different channels to RT and NRT users. On the contrary, this chapter aims to provide *access of same channel to both RT and NRT users*, thus increasing system heterogeneity on a per-channel basis.

(iii) Literature Survey: VoIP Applications in CRN

Establishing VoIP in CRN is an emerging area of research. VoIP capacity is analyzed using DTMC in [6.35], that is extended with energy detection based spectrum sensing in [6.36] and with VoIP traffic on multiple channels in [6.37]. VoIP capacity is also studied under channel access schemes in [6.38] and scheduling algorithms in [6.39], along with design of MAC layer based on-demand sensing in [6.40]. VoIP traffic is further analyzed with ON-OFF and Poisson Models in [6.41], that is extended in [6.42] by including effects of imperfect spectrum sensing. Joint packet and connection-level analysis based call admission control policies are also designed to increase VoIP Erlang capacity in [6.43] and by including Primary Resource Occupancy information in [6.44]. However, these works do not consider advanced VoIP features such as codecs, call signaling protocols, etc. On the contrary, the work in this chapter uses *VoIP silence suppression* to design CRN with higher spectrum efficiency.

The novelty of the work in this chapter is ascertained by the fact that the proposed VoIP based 2-tier CRN aims to increase system capacity by allowing two tiers of heterogeneous SUs in the same channel after taking advantage of silence suppression features in VoIP calls. To the best of our knowledge, no such work on 2-tier CRN has been reported so far, as every research has been directed towards a single-tier based CRN.

6.1.2 Significant Contributions

The primary contribution of this work is to design and implement VoIP based 2-tier CRN that allows two SUs to operate concurrently on the same channel. The significant aspects of this work can be summarized as follows.

1. The novelty of the proposed model for VoIP based 2-tier CRN is established following an in-depth literature survey in Section 6.1.
2. The system model is developed in Section 6.2 along with necessary design considerations.
3. The analytical models for VoIP based 2-tier CRN are designed and analyzed for performance efficiency over the Basic one-tier CRN in Section 6.3.
4. Markov Models are designed step-wise in Section 6.4 to map the interaction between primary and secondary users in both tiers by including suitable traffic distribution models and system parameters along with the incorporation of spectrum handoff operations.
5. The developed model is validated for performance improvement by implementing a real-world VoIP based 2-tier CRN under OPNET Modeler 16.0.A in Section 6.5.

6. The practical issues involved in designing VoIP based 2-tier CRN are discussed in Section 6.6 with the design and implementation of two algorithms based on message passing policy with the aim of increasing system capacity while reducing interference among the participating users.
7. A real prototype of the model is designed in hardware test-bed and its practical significance is studied with respect to disaster management systems in Section 6.7.

Finally, the chapter is concluded in Section 6.8.

6.2 System Model Design for 2-Tier CRN

In a Basic CRN model comprising of PUs and SUs, every PU is allotted a licensed spectrum band where it performs transmission for a certain duration. However, a PU may not transmit continuously. SU senses the band for PU presence in the sensing time interval (denoted by t_s). If the PU is sensed absent, the SU starts its transmission in that band for the secondary transmission time interval (denoted by t_d) [6.11]. The Basic model is depicted in Fig. 6.1.

On contrary to the single-tier based Basic CRN, the work in this chapter proposes a VoIP based 2-tier CRN that allows two tiers of SUs to transmit in the absence of PUs. The principle behind this design is that VoIP transmission occurs in the form of talkspurts and consumes 40-50 % of the total transmission time on an average [6.9], [6.45]. Hence, VoIP SUs do not completely utilize the secondary transmission time slot (t_d). Accordingly, other SUs that are involved in data communication can be incorporated in the system to utilize t_d time slot when VoIP SUs are idle.

6.2.1 Proposed Model

The model design for VoIP based 2-tier CRN broadly encompasses 3 basic processes and are discussed as follows.

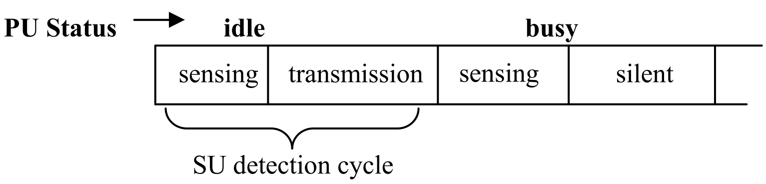


Fig. 6.1 Working Model of Secondary User in Basic CRN

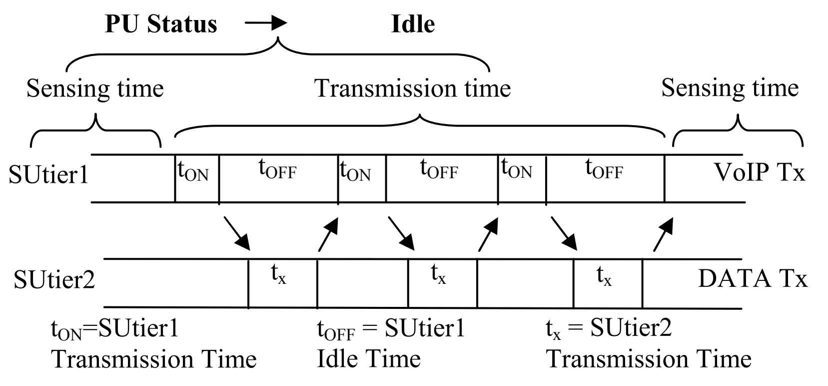


Fig. 6.2 Design of VoIP based 2-tier CRN

1. *Traffic Classification*: SUs are classified as VoIP (real-time) and DATA (non real-time) SUs that are involved in VoIP and DATA communication respectively. This classification is done by spectrum controller node [6.1] that coordinates between the VoIP and DATA Servers.
2. *1st-tier SU Allocation*: As per the principle of 2-tier CRN Model, VoIP SU accesses idle channel in the absence of any PU in that channel for t_d time slot. DATA SU is not allowed to occupy this idle channel even if it is ready for transmission. The priority of VoIP SUs over DATA SUs is supported by the fact that Quality of Service (QoS) provisioning for VoIP demands higher priority for VoIP SUs in accessing idle channels.

Therefore, VoIP SU constitutes the 1st-tier of the 2-tier CRN and is denoted by SUtier1.

3. *2nd-tier SU Allocation*: DATA SU accesses the same channel that has been used by VoIP SU (SUtier1) in t_d time interval when VoIP SU is in idle mode and has implemented the silent suppression with the help of its codec. The access of the channel to the DATA SU is granted by the spectrum controller node.

DATA SU, therefore, constitutes the 2nd tier of CRN and is denoted by SUtier2. The model is represented in Fig. 6.2.

It must be noted that in VoIP based 2-tier CRN, SUtier1 can always reclaim the channel back from SUtier2 with the help of spectrum broker. This design must be clearly distinguished from the concept of statistical multiplexing

[6.46] used frequently in VoIP based systems. This system does not implement statistical multiplexer due to several practical difficulties. The main problem lies in implementing centralized statistical multiplexer for controlling SU transmissions in overlay CRN, since SUs have primary access for transmission in idle channels. Moreover, during spectrum handoff, SUs sense for idle frequency bands and reorient their transceivers accordingly using their cognitive capabilities. This spectrum mobility is not supported by a normal statistical multiplexer. These problems are illustrated in the test-bed implementation of 2-tier CRN in Section 6.7.

6.2.2 Design Considerations

Successful implementation of the developed model is subject to certain conditions that are considered to be fulfilled in this study. Firstly, VoIP applications are deemed to have been effectively implemented in the 1st tier of CRN as discussed thoroughly in [6.35], [6.36]. This consideration is justified because this work does not deal with the complexities of VoIP transmission in CRN (which have been discussed already in the previous chapters) but has an entirely different objective. Moreover, VoIP SUs must implement codecs supporting silent suppression and Voice Activity Detection (VAD) techniques. As focus is on utilizing silence periods, individual codecs are not the point of concern in this work and are not explored.

The system is considered to be comprised of large number of VoIP and DATA SUs so that at any point of time, they are always available to occupy the 1st and 2nd tier of CRN respectively. Suitable MAC protocols are to be implemented in this regard to select the VoIP SUs for the 1st tier and DATA SUs for the 2nd tier of CRN. Selection of MAC protocols depend on several factors like the overall system architecture (centralized or distributed), handoff policies (proactive or reactive), etc. Since this work establishes the system from the spectrum utilization point of view, analysis of MAC protocols is not performed here.

Finally we consider that every occupied channel is released by SUtier2 on request of SUtier1 at a certain time prior to onset of either a new talkspurt for

SUtier1 or end of SUtier1 transmission time slot. This entire process incurs a delay that is indicated by inclined arrows in Fig 6.2.

6.3 Mathematical Modeling and Analysis of VoIP Based 2-Tier CRN

This section deals with the formulation of VoIP traffic in a wireless channel and subsequent design and analysis of mathematical models for VoIP based 2-tier CRN. The primary reason of this mathematical formulation is to establish the trade-off between increasing the system capacity without hampering PU transmissions. One of the critical aspects of system design lies in optimally configuring the involved parameters and this section achieves this task through extensive mathematical modeling and inferences derived thereafter.

6.3.1 Formulation of VoIP Traffic in a Wireless Channel

A wireless channel is designed as a two-state Markov process as already explained in Chapter 5. Accordingly, the transition probability that there are *m* unoccupied channels in the current frame and *n* unoccupied channels in the next frame is given by (5.1).

As network conditions vary with time, an *event driven* approach is followed in this work where each event represents a particular time interval 't' and is expressed as,

$$t = t_s + t_d \tag{6.1}$$

where t_s and t_d denote SU sensing and transmission time intervals respectively. Henceforth, design of every parameter in this chapter follows the event driven approach based on 't'.

Let $p_r(t)$ be the probability of PU arrival in a particular channel at time interval t. When each of the M wireless channels is designated to a single PU, the total number of available unoccupied channels at time interval t is given by,

$$m(t) = M(1 - p_r(t)) \tag{6.2}$$

However, due to imperfect spectrum sensing by SU, the total number of measured unoccupied channels is derived as,

$$m'(t) = m(t) - m(t) \times p_f(\cdot) + (M - m(t)) \times (1 - p_d(\cdot)) \tag{6.3}$$

where p_f and p_d denote probability of false alarm and probability of detection respectively.

SUtier1 transmits during the talkspurt period and remains idle during silent suppression interval with probabilities P_{busy} and P_{idle} respectively. Accordingly, VoIP traffic for SUtier1 is formulated as an on-off model with $1/\alpha$ and $1/\beta$ as on and off periods respectively, that are exponentially distributed. Let $R_q(V)$ be VoIP transmission rate for SUtier1 at idle channel q. The average VoIP transmission rate is 50 or 33.3 packets per second (pps) and depends on codecs implemented [6.47]. The throughput of SUtier1 in t_d time in channel q is expressed as,

$$C_{SUtier\ 1} = R_q(V) \times t_d \times P_{busy}$$

where $$P_{busy} = \alpha^{-1} / (\alpha^{-1} + \beta^{-1}) \quad , \quad q \in m'(t) \tag{6.4}$$

Duration for which SUtier1 is silent in t_d time slot is given by,

$$T_{idle} = t_d \times P_{idle} \text{ where } P_{idle} = 1 - P_{busy} \tag{6.5}$$

The objective is to allow SUtier2 utilize T_{idle} by exploiting the silence suppression feature of VoIP codec in SUtier1 and is decided based on a binary variable P_{ss} that is defined by,

$$P_{ss} = 1 \ \forall \ Tidle - (t_{grant} + t_{reclaim}) > 0 \ ,$$
$$P_{ss} = 0 \ \forall \ Tidle - (t_{grant} + t_{reclaim}) < 0 \tag{6.6}$$

where, t_{grant} and $t_{reclaim}$ indicate total time incurred in granting channel access to SUtier2 and reclaiming it back by SUtier1 respectively. t_{grant} primarily depends on user access strategies as guided by MAC protocols. $t_{reclaim}$ is governed by SUtier1 talkspurt characteristic, transmission interval and functional specifications of spectrum controller node. Hence, analytical expressions for t_{grant} and $t_{reclaim}$ are implementation-specific.

Thus, P_{ss} paves the way towards developing mathematical models for VoIP based 2-tier CRN.

6.3.2 Mathematical Modeling of 2-Tier CRN for Identical SU Transmission Probabilities

The mathematical model for 2-tier CRN is designed with respect to the system capacity in terms of Sum Goodput for the SU that is described in Chapter 5. Let N_s be the total number of SUs, each transmitting at the rate R with probability p. Let N_p be the total number of PUs that transmit with the same probability p. Accordingly, SU Sum Goodput for the Basic CRN is derived in (5.3).

It is observed from (5.3) that SU Sum Goodput is calculated for all possible combinations of SUs and PUs. However, event driven approach as already stated in Section 6.3.1 is used to derive SU Sum Goodput in this work because of the following reasons.

1. The assumption of equal transmission probabilities for both PUs and SUs as adopted in (5.3) is not justified as PUs and SUs are independent sets of users. With the event driven approach, it is possible to separately model both PU and SU arrival probabilities.
2. Unlike (5.3), event driven modeling makes it possible to model the PU arrival probability $p_r(t)$ corresponding to a certain traffic distribution with respect to every time interval 't' and therefore, capture the dynamic behavior of the PUs.

(i) <u>Modeling SUtier1 in 2-tier CRN</u>

Let total number of SUs in the CRN be denoted by Ns and defined as,

$$Ns = Ntier1 + Ntier2 \tag{6.7}$$

where, $Ntier1$ and $Ntier2$ denote the total number of SUtier1 and SUtier2 respectively.

Let SUtier1 be involved in VoIP communication for a total of T time intervals and transmits at $R_q(V)$ rate per second at an idle channel q. Let q belongs to $m'(t)$. It is considered that at any time interval t, j_{VoIP} number of SUtier1s arrives in CRN with probability p. Therefore, j_{VoIP} number of SUs is active with the probability $p^{j_{VoIP}}(1-p)^{Ns-j_{VoIP}}$. Hence, rewriting '$i$' in terms of $m'(t)$ and incorporating the time interval t in (5.3), the SU Sum Goodput for SUtier1 is expressed by (6.8).

$$C_{SUtier1}^{sum} = \sum_{t=1}^{T}\sum_{j=1}^{Ns}\binom{N_s}{j}p^{j_{VoIP}}(1-p)^{Ns-j_{VoIP}}\sum_{q=1}^{m'(t)}\left\{R_q\left(1-\frac{1}{m'(t)}\right)^{j_{VoIP}-1}\right\} \tag{6.8}$$

Hence, the number of SUs transmitting at a particular time interval 't' is given by,

$$j = Ntier\ 1_{\max} \times p \tag{6.9}$$

(ii) ***<u>Modeling SUtier2 in 2-tier CRN</u>***

SUtier2 is allowed to transmit only when P_{ss} parameter as defined by (6.6) for the corresponding SUtier1 is 1 (considering identical transmission characteristics of all SUtier1). Let $R_q(D)$ represents data transmission rate for SUtier2 at an idle channel q where q belongs to $m'(t)$, j_{DATA} denotes total number of SUtier2 that has arrived in CRN with probability p. At time interval 't', the throughput of SUtier2 per unit time in channel q is given by (6.10).

$$C_{SUtier\ 2}^{rate}(q,t) = P_{ss}R_q(D)\left(1-\frac{1}{m'(t)}\right)^{j_{Data}-1} \tag{6.10}$$

Thus, the final SU Sum Goodput for 2-tier CRN is derived by incorporating (6.10) in (6.8) and is expressed by (6.11). Let C_{2-TIER}^{sum} denote the SU Sum Goodput for 2-tier CRN. Therefore, we have

$$C_{2-TIER}^{sum} = \sum_{t=1}^{T}\sum_{j=1}^{Ns}\binom{N_s}{j}p^{j}(1-p)^{Ns-j}\sum_{q=1}^{m'(t)}\left\{R_q(V)\left(1-\frac{1}{m'(t)}\right)^{j_{VoIP}-1} + P_{ss}R_q(D)\left(1-\frac{1}{m'(t)}\right)^{j_{Data}-1}\right\}$$

$$\forall j = j_{VoIP} + j_{DATA} \tag{6.11}$$

(iii) Performance Evaluation

The 2-tier CRN model is hence analyzed for performance improvement over the Basic CRN. As Basic CRN corresponds to CRN comprising of SUtier1 only, SU Sum Goodput for Basic CRN is same as that of SUtier 1 as expressed in (6.8). Similarly, optimal number of SUs supported in Basic CRN (denoted by $Nbasic_{max}$) is equal to $Ntier1_{\max}$.

Normalized SU Sum Goodput for Basic and 2-tier CRN as derived in (6.8) and (6.11) respectively is plotted in Fig. 6.3 with respect to *Ntier1*. Rise in PU activity is modeled by increasing $p_r(t)$ as t varies from 1 to T. As a result, $m'(t)$ decreases symbolizing a network with high traffic load. Three scenarios are considered in Fig. 6.3, where the number of SUs in CRN is i) greater than, ii) equal to, and iii) less than the total number of idle channels.

Mathematically, these scenarios are represented as, $Ntier1 \times p > m'(t)$, $Ntier1 \times p = m'(t)$ and $Ntier1 \times p < m'(t)$ respectively.

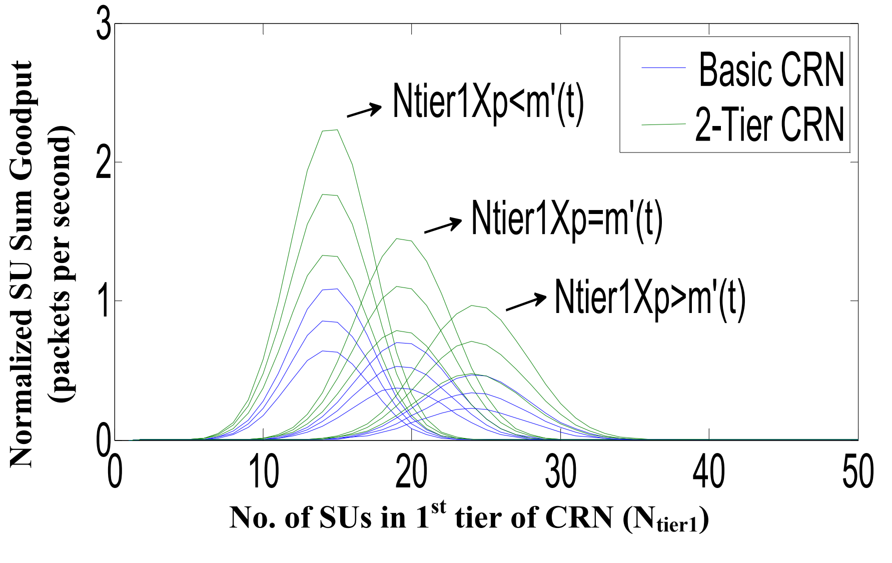


Fig. 6.3 Variation in Normalized SU Sum Goodput with total number of SUs in the first tier of CRN

It is observed in Fig. 6.3 that the SU Sum Goodput for 2-tier CRN is higher than that of Basic CRN in all the three scenarios. However, as *Ntier1* is increased, SU Sum Goodput decreases because the SU channel access probability decreases as more SUs compete for the same channel.

In Scenario 1 ($Ntier1 \times p < m'(t)$), the number of supported SUs in CRN decreases, representing system under-utilization in terms of user capacity. In Scenario 3 ($Ntier1 \times p > m'(t)$), SU Sum Goodput decreases due to increase in collision. As confirmed in Fig. 6.3, the trade-off between SU Sum Goodput and *Ntier1* is optimal in Scenario 2, as defined by the condition,

$$Ntier1 \times p = m'(t) \tag{6.12}$$

The validity of (6.12) is further established as follows.

Validation of the condition: $Ntier1 * p = m'(t)$

As per (6.12), $Ntier1 \times p = m'(t)$

$$= total\ \ number\ \ of\ \ measured\ \ idle\ \ channels \tag{6.13}$$

In the ideal scenario, the total no. of idle channels is expressed as,

$$m = M\,(1 - p_r) \tag{6.14}$$

where M = no. of channels in CRN, p_r = PU arrival probability.

Therefore, in an ideal scenario,

$$Ntier1 \times p = m \tag{6.15}$$

Considering $M = Np$, where Np=total number of PUs,

$$m = Np(1 - p_r) \tag{6.16}$$

Hence, from (6.15), $Ntier1 \times p = Np\,(1 - p_r)$ (6.17)

For identical transmission probabilities of PU and SU as denoted by p and considering total no. of SUs in Basic CRN = total no. of SUs in tier1 of 2-tier CRN ($Ntier1 = Ns$), (6.17) is modified as,

$$Ns \times p = Np(1 - p) \Rightarrow Ns = \frac{Np(1 - p)}{p} \tag{6.18}$$

Equation (6.18) has already been established in [6.8]. Since (6.18) is derived from (6.12), (6.12) is validated with respect to [6.8].

The optimal SU transmission probability (denoted by p^*) is derived from (6.12) and expressed as,

$$p^* = m'(t)/Ntier1 \tag{6.19}$$

Let the optimal number of SUs (that achieve maximum SU Sum Goodput) for Basic CRN and 2-tier CRN be denoted by $Nbasic_{max}$ and Ns_{max} respectively. Fig. 6.4 plots the variation in $Nbasic_{max}$ and Ns_{max} with SU transmission probability p. $Nbasic_{max\_observed}$ and $Ns_{max\_observed}$ are SUs in Basic and 2-tier CRN that yield maximum SU Sum Goodput as per (6.8) and (6.11) respectively. It is seen in Fig. 6.4 that $Ns_{max\_observed}$ closely resembles $Ns_{max\_calculated}$, that is defined by the following condition.

$$Ns_{max\_calculated} = Nbasic_{\max}(1+P_{ss})\ \forall p \neq 0 \tag{6.20}$$

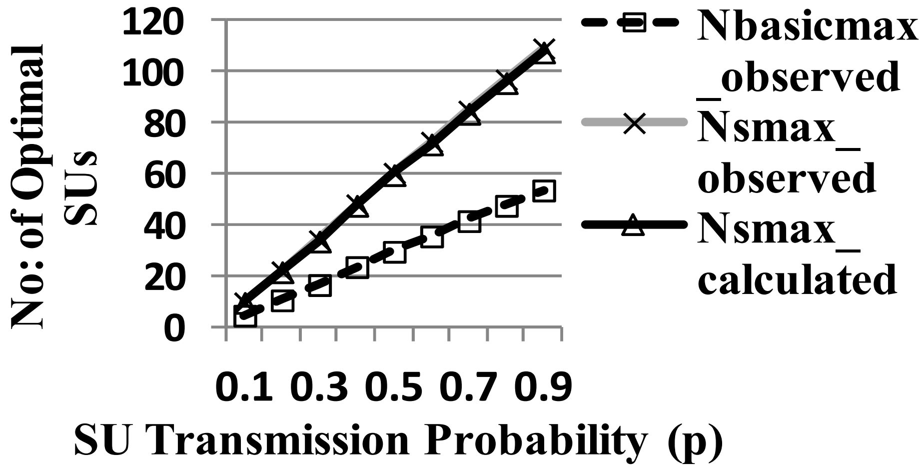


Fig. 6.4 Number of Optimal SUs versus their Transmission Probability, where $Nbasic_{max\_observed}$ and $Ns_{max\_observed}$ are SUs in Basic and 2-tier CRN that yield the maximum SU Sum Goodput as per (6.8) and (6.11) respectively and $Ns_{max\_calculated}$ is obtained according to (6.20)

Hence, the number of optimal SUs in 2-tier CRN, Ns_{max} increases by almost $(1+P_{ss})$ times as compared to the Basic CRN. The expression for Ns_{max} can also be derived using first order mathematical logic as discussed below.

Proof:

SUtier2 can transmit when P_{ss} value for SUtier1 is 1. Hence, optimum number of SUs in 2-tier CRN is given by,

$$Ns_{\max} \times p = Ntier1_{\max} \times p_{tier1} + Ntier2_{\max} \times p_{tier2} \quad (6.21)$$

For equal transmission probabilities of all SUs,

$$Ns_{\max} \times p = Ntier1_{\max} \times p + Ntier2_{\max} \times p$$

$$= \left(Nbasic_{\max} \times p\right) + \left(P_{ss}\right)\left(Nbasic_{\max} \times p\right)$$

(Since, $Ntier1_{max}$, $Ntier2_{max}$=$Nbasic_{max}$)

$$= Nbasic_{\max} \left(1 + P_{ss}\right) \forall p \neq 0 \quad (6.22)$$

where $Ntier1_{max}$, $Ntier2_{max}$ = maximum no. of SUtier1 and SUtier2 with arrival probabilities p_{tier1} and p_{tier2} respectively.

These results directly lead us to the following Proposition.

Proposition 6.1

For a VoIP based 2-tier CRN, optimal number of SUs having equal transmission probabilities increases by (1+P_{ss}) times than in case of the basic scenario. The optimal SU transmission probability in 2-tier CRN is obtained by dividing the number of idle channels by the total number of SUs.

It can be directly deduced from Proposition 1 that for identical transmission rates of both SUtier1 and SUtier2 and considering only a single transmission instance by SUtier2 in one idle slot of SUtier1, SU Sum Goodput records a minimum increase by 2 times in 2-tier CRN, as compared to the Basic CRN.

6.3.3 Mathematical Modeling of 2-Tier CRN for Varying SU Transmission Probabilities

The assumption of all SUs having equal transmission probabilities is relaxed to present a generalized scenario where every SU transmits with a different probability. It implies that all SUs may not arrive at the same time and are not equally likely to get access to a particular idle channel. Let p_i denote the transmission probability of i^{th} SU where each such SU denotes either SUtier1 or SUtier2. Intuitively, SUs with higher p_i are more likely to get an idle channel and vice-versa.

(i) Designing SUtier1 in 2-tier CRN

Let *U(v)* be the set of SUtier1s with decreasing value of p_i and is expressed as

$$U(v) = \{U1(v), U2(v),.., UNtier\ 1(v)\}$$

$$\forall\, p_{U1(v)} > p_{U2(v)} > .. > p_{UNtier\ 1(v)} \tag{6.23}$$

The throughput per unit time for SUtier1s in *U (v)* are given by,

$$C_{SUtier\_1}^{rate\_U1(v)}(t) = \left[R_1(V)\left(1 - \frac{1}{m'(t)}\right)^{j_{VoIP}-1} \right]_{U1(v)}$$

$$C_{SUtier\_1}^{rate\_U2(v)}(t) = \left[R_2(V)\left(1 - \frac{1}{m'(t)-1}\right)^{j_{VoIP}-2} \right]_{U2(v)}$$

$$\vdots$$

$$C_{SUtier\,1}^{rate\_Ui(v)}(t) = \left[R_i(V)\left(1 - \frac{1}{m'(t)-i+1}\right)^{j_{VoIP}-i} \right]_{Ui(v)} \tag{6.24}$$

(ii) Designing SUtier2 in 2-tier CRN

Let *U(d)* be the set of SUtier2 with decreasing value of p_i and is denoted by (6.25).

$$U(d) = \{U1(d), U2(d),.., UNtier\ 2(d)\}$$

$$\forall\, p_{U1(d)} > p_{U2(d)} > .. > p_{UNtier\ 2(d)} \tag{6.25}$$

Unlike the previous scenario in Section 6.3.2, all SUtier1 do not arrive with identical probability. When a certain SUtier1 is in idle mode, the decision to allow SUtier2 to access the channel depends on the transmission characteristics of that SUtier1 on that channel. Hence, P_{ss} in this scenario is dependent on the behavior of SUtier1 in a particular channel and hence is expressed as $P_{ss}(q)$ where q denotes the channel that is occupied by the corresponding SUtier1.

As per (6.25), $U1(d)$ is the first SUtier2 to arrive in CRN. It is given access to an idle channel q only when the corresponding $P_{ss}(q)$ is equal to 1. Otherwise, it moves to the next available idle channel $q+1$ and accesses it based on the value of $P_{ss}(q+1)$ and so on.

The throughput per unit time for $U1(d)$ is, therefore, expressed as

$$C_{SUtier\,2}^{rate\_U1(d)}(t) = \Bigg[\Bigg[\left\{P_{ss}(1)R_1(D)\left(1-\frac{1}{m'(t)}\right)^{j_{DATA}-1}\right\}$$

$$+\{1-P_{ss}(1)\}\left\{P_{ss}(2)R(D)\left(1-\frac{1}{m'(t)}\right)^{j_{DATA}-1}\right\} \vdots$$

$$+\prod_{r=1}^{m'(t)-1}\left(1-P_{ss}(r)\right)\left\{P_{ss}(m'(t))R(D)\right\}\Bigg]_{U1(d)} \qquad (6.26)$$

Thus, it is observed from (6.26) that there are several possible options with respect to channel allotment to SUtier2 based on the value of $P_{ss}(q)$ and these options are depicted in Fig. 6.5.

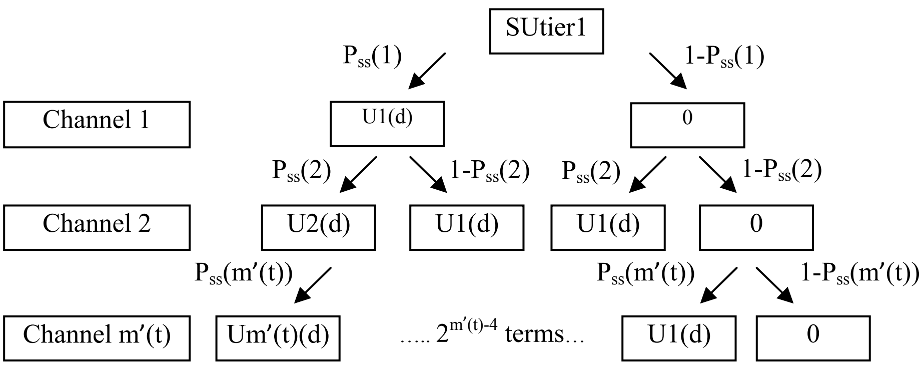


Fig. 6.5 Different Scenarios for Channel Allocation to SUtier2

Accordingly, the SU Sum Goodput for 2-tier CRN with varying probabilities is derived in (6.27).

$$C_{2-tier}^{sum1} = \sum_{t=1}^{T}\sum_{j=1}^{Ns}\binom{Ns}{j}\prod_{k=1}^{j}p_k\prod_{k=1}^{Ns-j}(1-p_k)\left[\left[R_1(V)\binom{m'(t)}{1}\left(\frac{1}{m'(t)}\right)\left(1-\frac{1}{m'(t)}\right)_{\forall U1(v)}^{j_{VoIP}-1}+\right.\right.$$

$$\left.P_{ss}(1)R_1(D)\binom{m'(t)}{1}\left(\frac{1}{m'(t)}\right)\left(1-\frac{1}{m'(t)}\right)_{\forall U1(d)}^{j_{DATA}-1}\right]$$

$$+\left[P_{ss}(1)\left\{R_2(V)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)_{\forall U2(v)}^{j_{VoIP}-2}+\right.\right.$$

$$\left.P_{ss}(2)R_2(D)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)_{\forall U2(d)}^{j_{DATA}-2}\right\}$$

$$+(1-P_{ss}(1))\left\{R_2(V)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)_{\forall U2(v)}^{j_{VoIP}-2}+\right.$$

$$\left.\left.P_{ss}(2)R_2(D)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)_{\forall U1(d)}^{j_{DATA}-1}\right\}\right]$$

$$+\ldots+\left[\prod_{r=1}^{m'(t)-1}P_{ss}(r)\left\{R_{m'(t)}(V)_{\forall Um'(t)(v)}+P_{ss}R_{m'(t)}(D)_{\forall Um'(t)(d)}\right\}+\ldots\right.$$

$$+(2^{m'(t)}-4)\ terms$$

$$\left.\left.+\prod_{r=1}^{m'(t)-1}(1-P_{ss}(r))\left\{R_{m'(t)}(V)_{\forall Um'(t)(v)}+P_{ss}R_{m'(t)}(D)_{\forall U1(d)}\right\}\right]\right]$$

$$\forall j = j_{VoIP} + j_{DATA} \tag{6.27}$$

(iii) *<u>Performance Evaluation</u>*

It is witnessed in Fig. 6.5 that there are $2^{m'(t)}$ number of possibilities for allocation of the last idle channel to SUtier2. Subsequently, maximum and minimum numbers of supported SUs in 2-tier CRN are derived accordingly.

1. *Maximum Number of Supported SUs:* The system supports maximum number of SUs only when every SUtier1 allows SUtier2 to transmit. This scenario corresponds to the leftmost child of the tree as shown in Fig. 6.5 and is defined by the following condition.

$$P_{ss}(q) = 1 \ \forall \ q \in m'(t) \tag{6.28}$$

It is imperative from (6.27) that the first term corresponding to the last idle channel satisfies the above condition and is given by,

$$R_{m'(t)}(V)_{\forall \ Um'(t)(v)} + P_{ss} R_{m'(t)}(D)_{\forall \ Um'(t)(d)}$$

where *Um′(t)(v)* and *Um′(t)(d)* refer to *m′(t)* th SUtier1 and *m′(t)* th SUtier2 respectively. It reveals that all previous SUtier1 in the *U(v)* set starting from *U1(v)* till *U(m′(t)-1)(v)* have been allotted idle channels and the same is applicable for SUtier2. Thus, in this scenario, *m′(t)* number of SUtier1 have been supported in the 2-tier CRN along with another *m′(t)* number of SUtier2. Hence, the maximum number of SUs supported in 2-tier CRN is expressed as.

$$\begin{aligned} Ns_{\max} &= (Ntier\,1 + Ntier\,2)_{\max} \\ &= m'(t) + m'(t) = 2\,Ntier\,1_{\max} \end{aligned} \tag{6.29}$$

2. *Minimum Number of Supported SUs:* In the worst case scenario, while every SUtier1 is allotted an idle channel, only one SUtier2 is allowed to access the channel. This scenario is represented by the leftmost child of the rightmost subtree in Fig. 6.5. The following condition depicts this scenario.

$$\exists_1 q \in m'(t) \mid P_{ss}(q) = 1 \tag{6.30}$$

The above condition is satisfied by the last term corresponding to the last idle channel in (6.27) and is given by,

$$R_{m'(t)}(V)_{\forall \ Um'(t)(v)} + P_{ss} R_{m'(t)}(D)_{\forall \ U1(d)}$$

It is clear that only 1 SUtier2 as denoted by *U1(d)* is allowed to transmit in an idle channel. However, *m′(t)* number of SUtier1 are still supported by the system. Hence, the minimum number of SUs supported in 2-tier CRN is expressed as,

$$Ns_{\min} = (Ntier\,1 + Ntier\,2)_{\min}$$

$$= m'(t) + 1 = Ntier1_{\max} + 1 \tag{6.31}$$

As $Nbasic_{max}$ is equal to $Ntier1_{max}$, it can be inferred from (6.29) and (6.31) that the optimal number of SUs supported by 2-tier CRN (Ns_{max}) for different SU transmission probabilities is given by the following relation.

$$Nbasic_{\max} + 1 \le Ns_{\max} \le 2Nbasic_{\max} \tag{6.32}$$

3. *Optimal SU Transmission Probability*: Let every SUtier1 and SUtier2 transmit with probabilities $p_{SUtier1}$ and $p_{SUtier2}$ respectively. Accordingly, the normalized SU Sum Goodput as per (6.27) is plotted with variation in *Ntier1*. While $p_{SUtier1}$ is kept fixed, $p_{SUtier2}$ is varied to obtain three patterns as observed in Fig. 6.6.

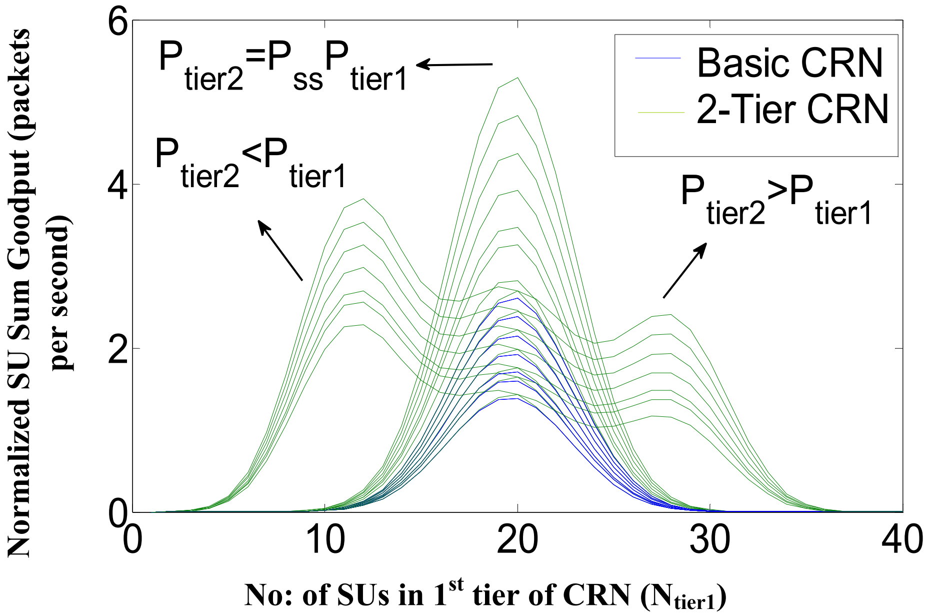


Fig. 6.6 Variation in Normalized SU Sum Goodput with total number of SUs in the first tier of CRN for different transmission probabilities of SUtier2

Figure 6.6 clearly denotes that the product of SU Sum Goodput and optimal number of SUs is highest under the following condition.

$$p_{SUtier\ 2} = P_{ss}\, p_{SUtier\ 1} \tag{6.33}$$

When all SUs in tier1 have different transmission probabilities, (6.33) can be generalized as,

$$p_{Ui(d)} = P_{SS}\, p_{Uj(v)} \tag{6.34}$$

Where, *Ui(d)* is the SUtier2 that accesses idle time slots in the channel occupied by *Uj(v)*.

All these results lead us to the following Proposition.

Proposition 6.2

In VoIP based 2-tier CRN, the optimal number of SUs with different transmission probabilities lies in the interval ($Nbasic_{max}$+1,2$Nbasic_{max}$) where $Nbasic_{max}$ denotes the optimal number of SUs supported by Basic CRN. Optimal transmission probability of each SUtier2 is given by P_{ss} times the optimal transmission probability of corresponding VoIP SUtier1.

6.4 Design and Analysis of Markov Model

After obtaining the different expressions for system capacity and transmission probabilities, the interaction among PU, SUtier1 and SUtier2 under diverse channel conditions is studied next in this section with the design of Markov Models. Markov Model serves as an effective tool to design CRN and has been implemented widely in recent works [6.5], [6.48]. The primary advantage of developing CRN with Markov Model is that it incorporates user-defined traffic distribution for PUs and SUs, along with customized network conditions and thereby, facilitates study of the complex interaction between PUs and SUs in the CRN.

Accordingly, this section designs Markov Models for Basic and 2-tier CRN and analyzes the increase in system capacity of 2-tier CRN over Basic

CRN with respect to SU dropping, blocking, handoff and transmission probabilities. The network is modeled as a collection of states where each state denotes channel status with respect to PU, SUtier1 and SUtier2. Let the steady state probability for every such state be denoted by P (i,j,k,l,m) where

- i= total number of active PUs transmitting in CRN,
- j= total number of active SUtier1 in CRN, that has arrived in the CRN,
- k= total number of active SUtier2 in CRN, that has been accepted by SUtier1,
- l=current "status" of SUtier1, and
- m=current "status" of SUtier2.

The term "status" denotes the action taken by SU under different network conditions. The various status symbols along with their meanings are described in Table 6.1.

Table 6.1 Status Symbols used in Markov Model

Status Value	Meaning	Definition
0	Transmission Mode	The SU has obtained access to a channel and is successfully transmitting.
	Null Mode	SU is not performing any transmission, handoff, blocking or dropping functions.
1	Handoff Mode	On PU arrival, the SU is performing spectrum handoff considering that an idle channel is available in the system. SU transmission is suspended temporarily during the handoff process.
2	Dropping Mode	SU transmission is suspended permanently as PU has arrived in the current channel and there are no idle channels available in CRN.
3	Blocking Mode	The incoming SU is not allowed to gain access to any channel for initiating transmission as there is no idle channel left in CRN.

Development of Markov model for 2-tier CRN is carried out incrementally in three phases. Initially, the first tier of CRN is modeled considering appropriate traffic distributions of PU and SUtier1. Secondly, SUtier2 is incorporated into the designed model following the principle of 2-tier CRN as discussed in Section 6.2. Finally, spectrum handoff is incorporated for all SUs in the CRN.

6.4.1 Markov Model Design for First Tier of CRN with Spectrum Handoff

Initially, the first tier of CRN is designed using Markov Model. It is obvious that in the absence of any further tier of SUs in the network, the first tier of CRN corresponds to the Basic CRN comprising of PUs and a single set of SUs. It is considered that PU and SUtier1 arrive in CRN following Poisson distribution with mean rates λ_p and λ_s respectively and have negative exponential service time distribution with mean rates $1/\mu_p$ and $1/\mu_s$ respectively. In order to design the Markov Model, P(i,j,k,l,m) is calculated for every possible state. As SUtier2 is not present, k = 0 for all P(•) in this scenario.

Spectrum handoff [6.4] is implemented for SUtier1 such that on arrival of PU in the current channel, SUtier1 shifts to available idle channel. It is to be noted that the implementation of spectrum handoff is dependent on several factors that include the underlying MAC protocol, CRN architecture, handoff policies, etc. and hence, its discussion is beyond the scope of this work. The generalized Markov model for CRN comprising of N channels is developed in Fig. 6.7 followed by the balance equations guiding the transmission of SUtier1.

The balance equations governing the transmission of SUtier1 in the Markov Model for CRN are defined as follows.

1. $\boxed{i+j \le N-1, i=0}$:

$$(\lambda_p + \lambda_s + j\mu_s)P(i,j,k,l,m) = \lambda_s P(i,j-1,k,l,m) + \mu_p P(i+1,j,k,l,m) + (j+1)\mu_s P(i,j+1,k,l,m) \quad (6.35)$$

2. $\boxed{i+j \le N-1, i \ne 0}$:

$$(\lambda_p + \lambda_s + j\mu_s + i\mu_p)P(i,j,k,l,m) = \left\{\frac{N-(i-1)-j}{N-(i-1)}\right\}\lambda_p P(i-1,j,k,l,m) +$$

$$\lambda_s P(i,j-1,k,l,m)+P(i,j-1,k,1,0)+(i+1)\mu_p P(i+1,j,k,l,m)+(j+1)\mu_s P(i,j+1,k,l,m)$$

(6.36)

3. $\boxed{i+j=N,\ i=0}$:

$$(\lambda_p+\lambda_s+j\mu_s)P(i,j,k,l,m)=P(i,j,k,3,0)+\lambda_s P(i,j-1,k,0,0) \quad (6.37)$$

4. $\boxed{i+j=N,\ i\neq 0}$:

$$(\lambda_p+\lambda_s+j\mu_s+i\mu_p)P(i,j,k,l,m)=\left\{\frac{N-(i-1)-j}{N-(i-1)}\right\}\lambda_p P(i-1,j,k,l,m)+$$

$$\lambda_s P(i,j-1,k,l,m)+P(i,j-1,k,1,0)+P(i,j,k,2,0)+P(i,j,k,3,0) \quad (6.38)$$

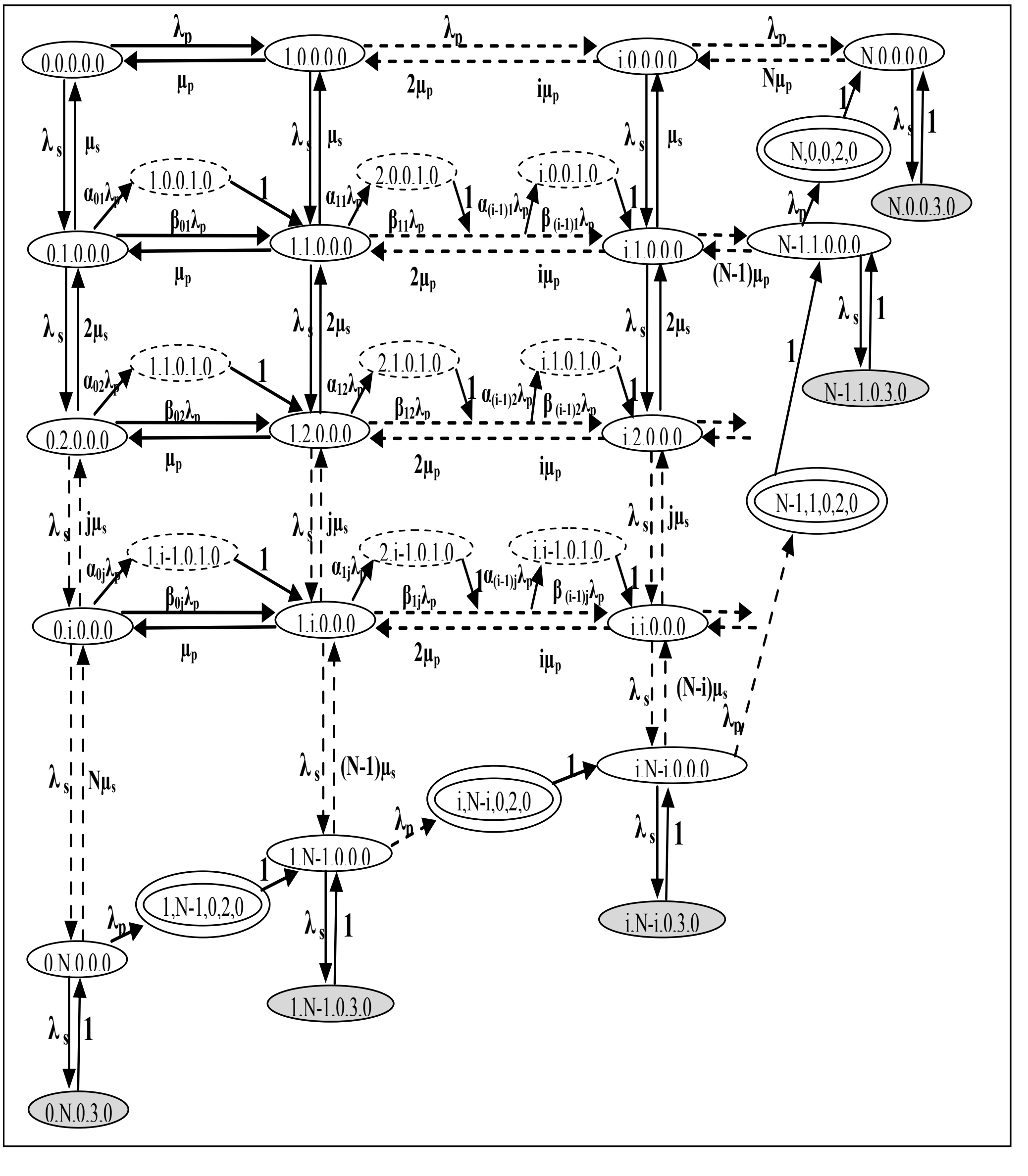


Fig. 6.7 Markov Model for the first tier of CRN

6.4.2 Markov Model Design for 2-Tier CRN with Spectrum Handoff only for SUtier1

In a 2-tier CRN, each SUtier1 allows SUtier2 to transmit during the silence periods as depicted in Fig. 6.2. Let SUtier2 arrive in CRN following Poisson distribution with λ_t as the mean rate and has negative exponential service time distribution with mean rate of $1/\mu_t$. Considering total number of PU and SUtier1 in the network at a certain time interval to be i and j respectively, the maximum number of SUtier2 admitted in CRN is j. The addition of SUtier2 by SUtier1 is depicted by a segment of the Markov Model in Fig. 6.8.

The maximum system capacity in terms of users admitted in 2-tier CRN is given by,

$$Cp_{\max} = PU + SUtier1 + SUtier2$$

$$= i + j + j = i + 2j \tag{6.39}$$

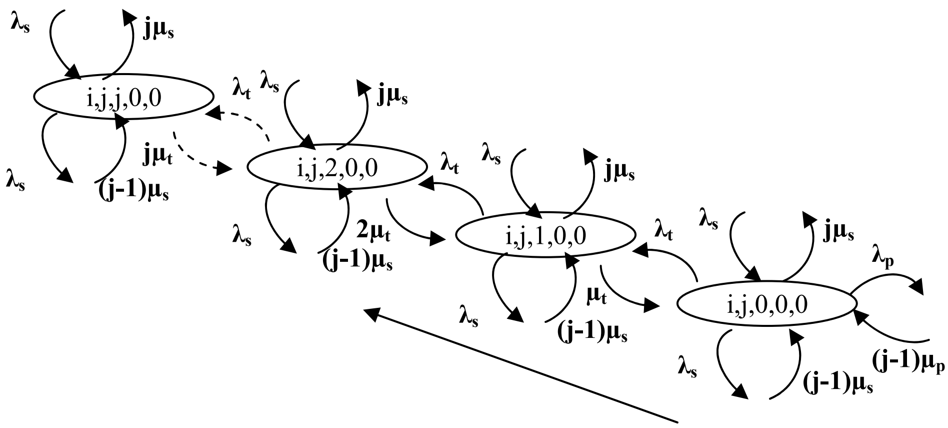


Fig. 6.8 Admission of SUtier2 in 2-tier CRN

SUtier2 does not perform spectrum handoff in this model. Rather, it is dropped under three conditions namely, i) at a time when SUtier1 is dropped, ii) when SUtier1 performs spectrum handoff, and iii) after SUtier1 finishes transmission. Therefore, the status of SUtier2, as denoted by "m" in P(i,j,k,l,m), accepts values of 0, 2 and 3 depending on its transmission, dropping and blocking mode respectively. At any point of time, status combinations for SUtier1 and SUtier2 as represented by {l,m} follow the conditions as described in Table 6.2.

Table 6.2 Status Conditions for the Designed Markov Model of 2-Tier CRN

Condition	Reason
$m = \{0,2\} \forall m \in \{l,m\} : l = 1$ $l = 1 \forall l \in \{l,m\} : m = 1$	SUtier2 is dropped when the corresponding SUtier1 implements spectrum handoff.
$m = \{0,3\} \forall m \in \{l,m\} : l = 3$ $l = 3 \forall l \in \{l,m\} : m = 3$	The fact that SUtier2 is blocked from accessing the channel implies that SUtier1 is already blocked.
$m = \{0,2\} \forall m \in \{l,m\} : l = 2$ $l = 0 \forall m \in \{l,m\} : m = 2$ when $j = k$ $= 2 \forall m \in \{l,m\} : m = 2$ when $j \neq k$	Both SUtier1 and SUtier2 transmissions can be dropped on the arrival of PU. A special case occurs when SUtier2 transmission is dropped when the transmission time interval for SUtier1 is over and the channel is released.

Accordingly, the Markov Model for 2-tier CRN (where spectrum handoff is performed by SUtier1 only) is illustrated in Fig. 6.9 along with the balance equations for SUtier1 and SUtier2.

The balance equations guiding the transmission of SUtier1 and SUtier2 in Markov Model for 2-tier CRN as per Fig. 6.9 are defined as follows.

CASE I: SUtier1

1. $\boxed{i+j \leq N-1, i=0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s)P(i,j,k,l,m) = \lambda_s P(i,j-1,k,l,m) + \mu_p P(i+1,j,k,l,m) + (j+1)\mu_s P(i,j+1,k,l,m) + \mu_t P(i,j,k+1,l,m) \quad (6.40)$$

2. $\boxed{i+j \leq N-1, i \neq 0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s + i\mu_p)P(i,j,k,l,m) = \beta_{(i-1)jk}\lambda_p P(i-1,j,k,l,m) + P(i,j-1,k,1,0) + \mu_t P(i,j,k+1,l,m) + \lambda_s P(i,j-1,k,l,m) + (i+1)\mu_p P(i+1,j,k,l,m) + (j+1)\mu_s P(i,j+1,k,l,m) + P(i,j-1,k,1,2) \quad (6.41)$$

3. $\boxed{i+j = N, i=0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s)P(i,j,k,l,m) = P(i,j,k,3,0) + \lambda_s P(i,j-1,k,l,m) + \mu_t P(i,j,k+1,l,m) \quad (6.42)$$

4. $\boxed{i+j = N, i \neq 0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s + i\mu_p)P(i,j,k,l,m) = \beta_{(i-1)jk}\lambda_p P(i-1,j,k,l,m) + \lambda_s P(i,j-1,k,l,m)$$

$$+P(i,j-1,k,1,0)+\mu_t P(i,j,k+1,l,m)+P(i,j,k,2,0)+P(i,j,k,3,3)+P(i,j-1,k,1,2) \tag{6.43}$$

CASE II: SUtier2

5. $\boxed{i+j\le N-1, k<j}$:

$$(\lambda_p+\lambda_s+\lambda_t+j\mu_s+k\mu_t)P(i,j,k,l,m)=(k+1)\mu_t P(i,j,k+1,l,m)+\lambda_s P(i,j-1,k,l,m)+$$
$$\mu_p P(i+1,j,k,l,m)+\lambda_t P(i,j,k-1,l,m)+(j+1)\mu_s P(i,j+1,k,l,m) \tag{6.44}$$

6. $\boxed{i+j\le N-1, k=j, i=0}$:

$$(\lambda_p+\lambda_s+j\mu_s+k\mu_t)P(i,j,k,l,m)=(j+1)\mu_s P(i,j+1,k,l,m)+$$
$$\mu_p P(i+1,j,k,l,m)+\lambda_t P(i,j,k-1,l,m)+P(i,j,k,0,2) \tag{6.45}$$

7. $\boxed{i+j\le N-1, k=j, i\ne 0}$:

$$(\lambda_p+\lambda_s+i\mu_p+j\mu_s+k\mu_t)P(i,j,k,l,m)=\delta(i-1)jk P(i-1,j,k,l,m)+(i+1)\mu_p P(i+1,j,k,l,m)+$$
$$\lambda_t P(i,j,k-1,l,m)+(j+1)\mu_s P(i,j+1,k,l,m)+P(i,j,k,0,2) \tag{6.46}$$

8. $\boxed{i+j=N, k<j, i\ne 0}$:

$$(\lambda_p+\lambda_s+\lambda_t+i\mu_p+j\mu_s+k\mu_t)P(i,j,k,l,m)=P(i,j-1,k,1,0)+P(i,j-1,k-1,1,2)+$$
$$\delta(i-1)jk\lambda_p P(i-1,j,k,l,m)+P(i,j,k,2,2)+P(i,j,k,2,0)+\lambda_s P(i,j-1,k,0,0)+$$
$$\lambda_t P(i,j,k-1,0,0)+(k+1)\mu_t P(i,j,k+1,0,0)+P(i,j,k,3,0) \tag{6.47}$$

9. $\boxed{i+j=N, k=j, i\ne 0}$:

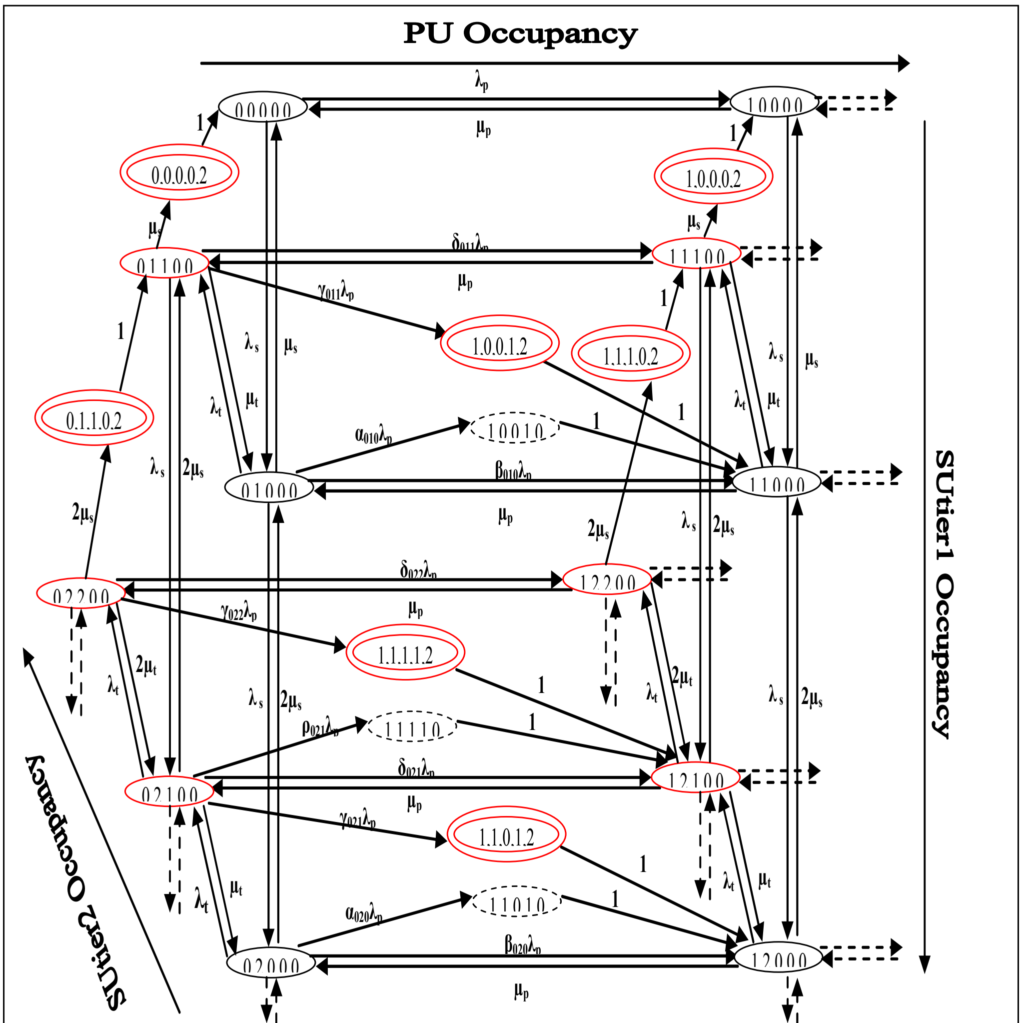
$$(\lambda_p+\lambda_s+i\mu_p+j\mu_s+k\mu_t)P(i,j,k,l,m)=\delta(i-1)jk\lambda_p P(i-1,j,k,l,m)+$$
$$P(i,j,k,2,2)+P(i,j,k,2,0)+\lambda_t P(i,j,k-1,l,m)+P(i,j,k,3,3) \tag{6.48}$$

10. $\boxed{i+j=N, k=j, i=0}$:

$$(\lambda_p+\lambda_s+j\mu_s+k\mu_t)P(i,j,k,l,m)=P(i,j,k,3,3)+\lambda_t P(i,j,k-1,l,m) \tag{6.49}$$

11. $\boxed{i+j=N, k<j, i=0}$:

$$(\lambda_p+\lambda_s+\lambda_t+j\mu_s+k\mu_t)P(i,j,k,l,m)=\lambda_s P(i,j-1,k,l,m)+\lambda_t P(i,j,k-1,l,m)+$$
$$(k+1)\mu_t P(i,j,k+1,l,m)+P(i,j,k,3,0) \tag{6.50}$$



User	State	Symbol	Expressions for Probabilities
SUtier1	Transmission	i,j,k,l,m	$\alpha_{ijk} = \frac{j}{N-i}$, $\beta_{ijk} = 1-\alpha_{ijk}$
	Handoff	i,j,k,l,m	$\gamma_{ijk} = \frac{k}{N-i}$, $\delta_{ijk} = 1-\gamma_{ijk}$ $\forall \ j=k$
SUtier2	Transmission	i,j,k,l,m	$\gamma_{ijk} = \frac{k}{N-i}$, $\rho_{ijk} = \frac{j-k}{N-i}$, $\delta_{ijk} = 1-(\gamma_{ijk}+\rho_{ijk})$ $\forall \ j \neq k$
	Dropping	i,j,k,l,m	

Fig. 6.9 Markov Model for 2-tier CRN with spectrum handoff only for SUtier1

6.4.3 Design of Markov Model for 2-Tier CRN with Spectrum Handoff for SUtier1 and SUtier2

In this section, Markov Model is designed for 2-tier CRN where both SUtier1 and SUtier2 perform spectrum handoff on sudden PU arrival. As PU arrives in the current channel and SUtier1 shifts to another channel, it sends information about the new channel to SUtier2. Thereafter, SUtier2 reorients its transceiver to the frequency band corresponding to new channel and, thus, implements spectrum handoff. However, it must be noted that since admission of SUtier2 in CRN is completely governed by SUtier1, spectrum handoff can be performed by SUtier2 only when corresponding SUtier1 executes spectrum handoff and is represented by the following condition.

$$m = \{0,1\} \forall m \in \{l, m\} : l = 1 \tag{6.51}$$

where *l, m* denote the status symbols in P(•).

Enabling handoff for all SUs in the network implies that as long as there are idle channels available in the system, the average system capacity is close to the maximum system capacity that is expressed in (6.39). Table 6.3 illustrates the conditions under which spectrum handoff can be performed by either only SUtier1 or both SUtier1 and SUtier2.

Table 6.3 Different Handoff And Dropping Conditions for SUtier1 and SUtier2

Condition	PU Arrival Status	Probability	Handoff by SUtier1 only		Handoff by SUtier1 and SUtier2	
			Value of {l,m}	*Remark*	*Value of {l,m}*	*Remark*
$j \neq k$	PU arrives at a channel occupied by both SUtier1 and SUtier2. There are idle channels available in CRN.	$\gamma = \frac{j}{N-i}$	{1,2}	Handoff by SUtier1. SUtier2 is dropped.	{1,1}	Handoff by SUtier1 and SUtier2
	PU arrives at a channel that is used by SUtier1 only. There are idle channels available in CRN.	$\rho = \frac{j-k}{N-i}$	{1,0}	Handoff only by SUtier1. SUtier2 is unaffected.	{1,0}	Handoff only by SUtier1. SUtier2 is unaffected.
	PU occupies the channel not used by both SUtier1 and SUtier2.	$\delta = 1-(\gamma+\rho)$	{0,0}	No handoff required	{0,0}	No handoff required
$j = k$	PU arrives at a channel occupied by both SUtier1 and SUtier2. There are idle channels available in CRN.	$\gamma = \frac{k}{N-i}$	{1,2}	Handoff by SUtier1. SUtier2 is dropped.	{1,1}	Handoff by SUtier1 and SUtier2
	PU occupies the channel not used by both SUtier1 and SUtier2.	$\delta = 1-\gamma$	{0,0}	No handoff required	{0,0}	No handoff required

However, handoff mechanisms fail when all the idle channels are occupied by PUs and SUs. Mathematically, it is represented by,

$$N = i + j \tag{6.52}$$

where N, i, j denote the total number of channels, PU and SUtier1 in CRN respectively.

In this scenario, it can be ascertained from Table 6.3 that,

$$\begin{aligned} \delta &= 1 - (\gamma + \rho) \\ &= N - (i + j) = 0 \end{aligned} \tag{6.53}$$

Any further arrival of PU results in two cases.

1. Case 1: $k < j, N > (i + j)$

   Only SUtier1 is dropped as there is no SUtier2 in this channel. The probability of SUtier1 being dropped on PU arrival is given by,

$$\gamma = \frac{j}{N - i} \tag{6.54}$$

2. Case 2: $k = j, N > (i + j)$

   Both SUtier1 and SUtier2 are dropped on arrival of PU with probability as expressed in (6.55).

$$\rho = \frac{j - k}{N - i} \tag{6.55}$$

3. Case 3: $k = j, N = (i + j)$

   SUtier1 and SUtier2 are dropped as PU arrives with probability = 1.

The complete Markov Model for 2-tier CRN with spectrum handoff implemented by all the SUs is depicted in Fig. 6. 10. Symbols as used in Table 6.3 are applied to denote the states in Fig. 6.10.

The balance equations governing the transmission of SUtier1 and SUtier2 in 2-tier CRN as per Fig. 6.10 are defined as follows.

CASE I: SUtier1

1. $\boxed{i+j \le N-1, i=0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s)P(i,j,k,l,m) = \lambda_s P(i,j-1,k,l,m) + \mu_p P(i+1,j,k,l,m) + (j+1)\mu_s P(i,j+1,k,l,m) + \mu_t P(i,j,k+1,l,m) \tag{6.56}$$

2. $\boxed{i+j \le N-1, i \ne 0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s + i\mu_p)P(i,j,k,l,m) = \beta_{(i-1)jk}\lambda_p P(i-1,j,k,l,m) + P(i,j-1,k,1,0) + \mu_t P(i,j,k+1,l,m) + \lambda_s P(i,j-1,k,l,m) + (i+1)\mu_p P(i+1,j,k,l,m) + (j+1)\mu_s P(i,j+1,k,l,m) + P(i,j-1,k,1,2) \tag{6.57}$$

3. $\boxed{i+j = N, i=0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s)P(i,j,k,l,m) = P(i,j,k,3,0) + \lambda_s P(i,j-1,k,l,m) + \mu_t P(i,j,k+1,l,m) \tag{6.58}$$

4. $\boxed{i+j = N, i \ne 0}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s + i\mu_p)P(i,j,k,l,m) = \beta_{(i-1)jk}\lambda_p P(i-1,j,k,l,m) + \lambda_s P(i,j-1,k,l,m) + P(i,j-1,k,1,0) + \mu_t P(i,j,k+1,l,m) + P(i,j,k,2,0) + P(i,j,k,3,3) + P(i,j-1,k,1,2) \tag{6.59}$$

CASE II: SUtier2

5. $\boxed{i+j \le N-1, k<j}$:

$$(\lambda_p + \lambda_s + \lambda_t + j\mu_s + k\mu_t)P(i,j,k,l,m) = (k+1)\mu_t P(i,j,k+1,l,m) + \lambda_s P(i,j-1,k,l,m) + \mu_p P(i+1,j,k,l,m) + \lambda_t P(i,j,k-1,l,m) + (j+1)\mu_s P(i,j+1,k,l,m) \tag{6.60}$$

6. $\boxed{i+j \le N-1, k=j, i=0}$:

$$\left(\lambda_p+\lambda_s+j\mu_s+k\mu_t\right)P(i,j,k,l,m)=(j+1)\mu_s P(i,j+1,k,l,m)+ \mu_p P(i+1,j,k,l,m)+\lambda_t P(i,j,k-1,l,m)+P(i,j,k,0,2) \quad (6.61)$$

7. $\boxed{i+j \le N-1, k=j, i\neq 0}$:

$$\left(\lambda_p+\lambda_s+i\mu_p+j\mu_s+k\mu_t\right)P(i,j,k,l,m)=\delta(i-1)jk\,P(i-1,j,k,l,m)+ (i+1)\mu_p P(i+1,j,k,l,m)+\lambda_t P(i,j,k-1,l,m)+(j+1)\mu_s P(i,j+1,k,l,m)+ P(i,j,k,0,2)+P(i,j-1,k-1,1,1) \quad (6.62)$$

8. $\boxed{i+j=N, k<j, i\neq 0}$:

$$\left(\lambda_p+\lambda_s+\lambda_t+i\mu_p+j\mu_s+k\mu_t\right)P(i,j,k,l,m)=P(i,j-1,k,1,0)+P(i,j-1,k-1,1,1)+ \delta(i-1)jk\,\lambda_p P(i-1,j,k,l,m)+P(i,j,k,2,2)+P(i,j,k,2,0)+\lambda_s P(i,j-1,k,0,0)+ \lambda_t P(i,j,k-1,0,0)+(k+1)\mu_t P(i,j,k+1,0,0)+P(i,j,k,3,0) \quad (6.63)$$

9. $\boxed{i+j=N, k=j, i\neq 0}$:

$$\left(\lambda_p+\lambda_s+i\mu_p+j\mu_s+k\mu_t\right)P(i,j,k,l,m)=\delta(i-1)jk\,\lambda_p P(i-1,j,k,l,m)+ P(i,j,k,2,2)+P(i,j,k,2,0)+\lambda_t P(i,j,k-1,l,m)+P(i,j,k,3,3) \quad (6.64)$$

10. $\boxed{i+j=N, k=j, i=0}$:

$$\left(\lambda_p+\lambda_s+j\mu_s+k\mu_t\right)P(i,j,k,l,m)=P(i,j,k,3,3)+\lambda_t P(i,j,k-1,l,m) \quad (6.65)$$

11. $\boxed{i+j=N, k<j, i=0}$:

$$\left(\lambda_p+\lambda_s+\lambda_t+j\mu_s+k\mu_t\right)P(i,j,k,l,m)=\lambda_s P(i,j-1,k,l,m)+\lambda_t P(i,j,k-1,l,m)+ (k+1)\mu_t P(i,j,k+1,l,m)+P(i,j,k,3,0) \quad (6.66)$$

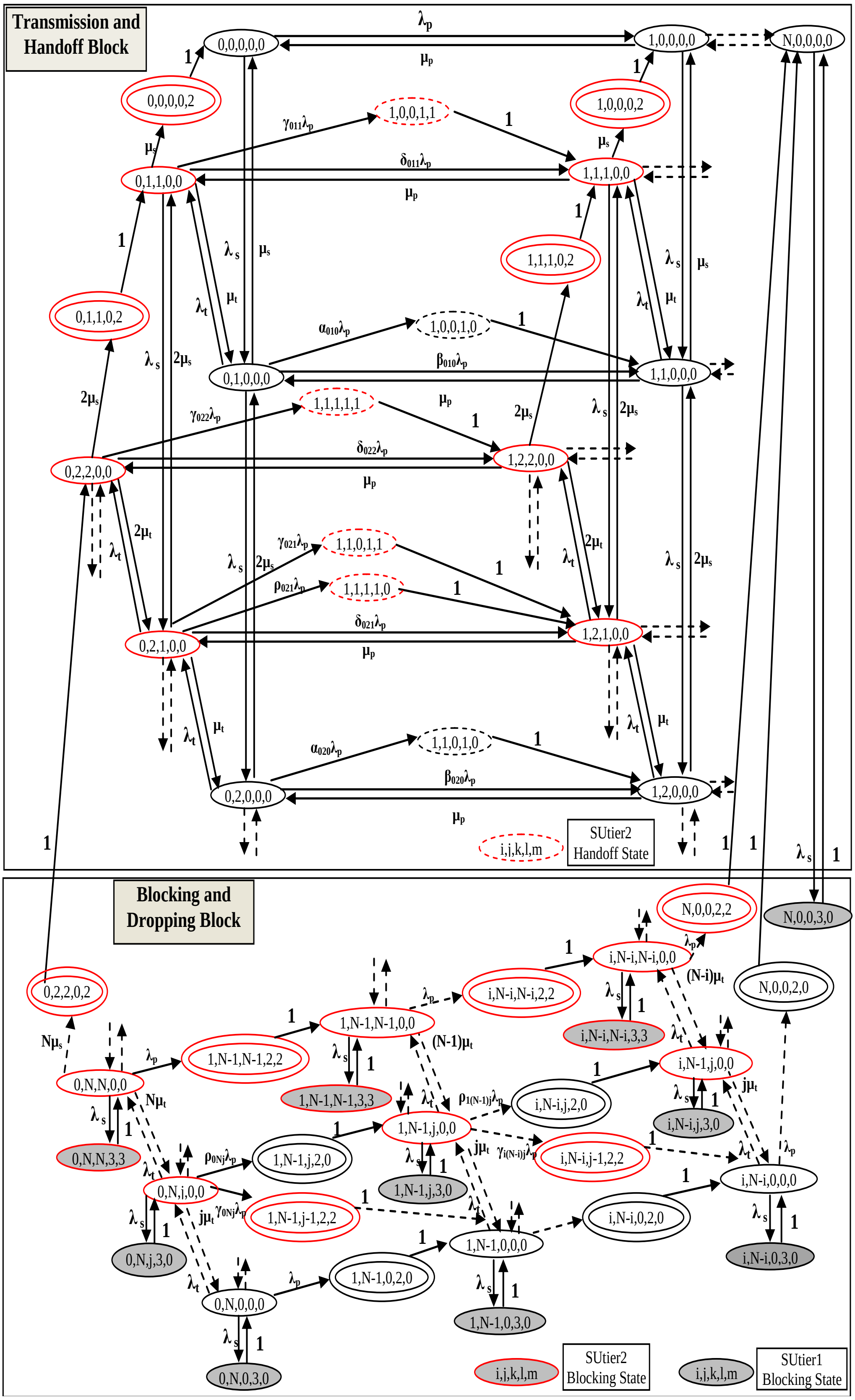


Fig. 6.10 Complete Markov Model for 2-tier CRN with spectrum handoff for both SUtier1 and SUtier2

6.4.4 Mathematical Formulation with Spectrum Handoff in 2-Tier CRN

A mathematical expression is derived to obtain the throughput for the 2nd SUtier1 that arrives in the proposed 2-tier spectrum handoff enabled CRN. Let $Ph_b(a)$ be the probability of spectrum handoff performed by bth SU in tier 1 to shift from the current channel a' to the a^{th} channel. Therefore, the throughput for the b^{th} SUtier1 having transmission rate $R_{a'}(VoIP)$ in a particular idle channel a' at a time interval t is derived from (6.24) and is by,

$$C_b^{SUtier\,1}(t) = (1 - Ph_b(a))\left\{ R_{a'}(VoIP)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)^{j-1}\right\}$$

(6.67)

Similarly, the throughput of SUtier2 with transmission rate as $R_{a'}(DATA)$ in the a' channel at time interval t is given by,

$$C_b^{SUtier\,2}(t) = P_{ss}(b)\left\{ R_{a'}(DATA)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)^{j-2}\right\}$$

(6.68)

Combining (6.67) and (6.68), the total throughput for a particular set of SUtier1 and SUtier2 is expressed in (6.69).

$$C_{SUthrou}(t) = (1 - Ph_b(a))\left[\left\{ Ra'(VoIP)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)^{j-1}\right\} + \right.$$

$$\left. P_{SS}(b)\left\{ Ra'(DATA)\binom{m'(t)-1}{1}\left(\frac{1}{m'(t)-1}\right)\left(1-\frac{1}{m'(t)-1}\right)^{j-2}\right\}\right] \quad (6.69)$$

There are several possibilities with respect to allotment of an idle channel to SUtier1 and is depicted as a three-layered tree in Fig. 6.11. For the 2nd SUtier1 in the system, the first layer determines whether the preceding SUtier1 grants access to SUtier2 or not. The second layer specifies the probabilities with which the 1st SUtier1 performs spectrum handoff in different channels. The third layer indicates the different spectrum handoff probabilities for the 2nd SUtier1.

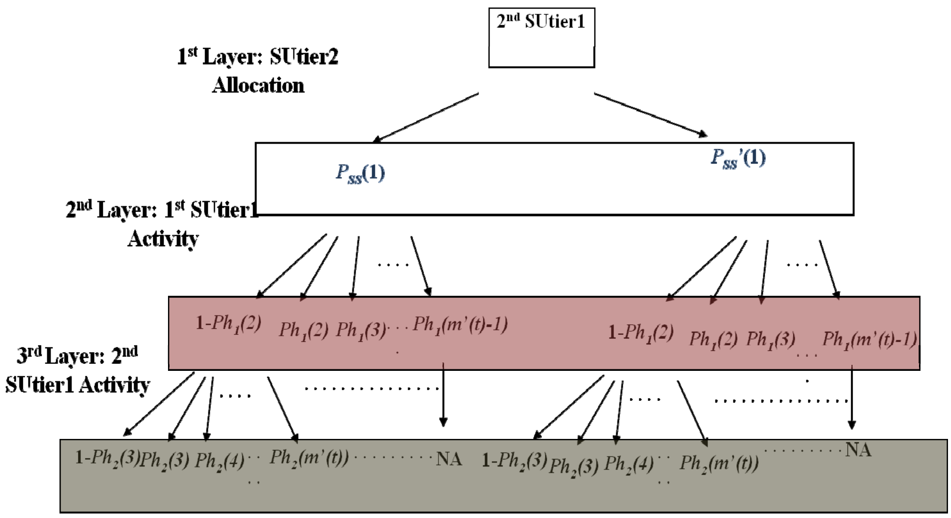


Fig. 6.11 Schematic Representation of all possibilities regarding channel allocation for 2nd SUtier1 on arrival in CRN

A special case occurs when the 1st SUtier1 performs repeated handoff and finally occupies one idle channel in the system which has only one more idle channel available. In this condition, the 2nd SUtier1 occupies this last available idle channel and is dropped on the event of any further PU arrival as it cannot perform any spectrum handoff. Let $C_{2ndSUthrou}$ denote the throughput of 2nd SUtier1 corresponding to the second layer of the tree. The general expression for $C_{2ndSUthrou}$ is derived in (6.68). It is further modified to include the different conditions of spectrum handoff as per Fig. 6.11 and is expressed in (6.71).

Let C_{2ndSU} be the overall throughput of 2nd SUtier1 at the topmost layer of the tree and is given by,

$$C_{2ndSU}(t) = P_{ss}(1)C_{2ndSUthrou}(t) + (1 - P_{ss}(1))C_{2ndSUthrou}(t) \quad (6.70)$$

$$C2ndSUthrou(t) = \left[\{(1 - Ph1(2))\} \left[(1 - Ph2(3)) \left[R2(VoIP) \binom{m'(t)-1}{1} \left(\frac{1}{m'(t)-1} \right) \left(1 - \frac{1}{m'(t)-1} \right)^{l} + \right. \right. \right.$$

$$\left. Pss(2)R2(DATA) \binom{m'(t)-1}{1} \left(\frac{1}{m'(t)-1} \right) \left(1 - \frac{1}{m'(t)-1} \right)^{m} \right]$$

$$+ \left[\sum_{k=3}^{m'(t)-1} Ph2(k) \left[Rk(VoIP) \binom{m'(t)-k+1}{1} \left(\frac{1}{m'(t)-k+1} \right) \left(1 - \frac{1}{m'(t)-k+1} \right)^{l} + \right. \right.$$

$$P_{ss}(2)R_k(DATA)\binom{m'(t)-k+1}{1}\left(\frac{1}{m'(t)-k+1}\right)\left(1-\frac{1}{m'(t)-k+1}\right)^m\Bigg]\Bigg]$$

$$+\left[Ph2\,(m'(t))\left[R_{m'(t)}\left(VoIP\right)+Pss\,(2)R_{m'(t)}\left(DATA\right)\right]\right]\Big]$$

$$+\Bigg[\left(Ph1(2)\right)\Bigg[\left(1-Ph2(4)\right)\Bigg[R3\left(VoIP\right)\binom{m'(t)-2}{1}\left(\frac{1}{m'(t)-2}\right)\left(1-\frac{1}{m'(t)-2}\right)^l$$

$$P_{ss}(2)R3\left(DATA\right)\binom{m'(t)-2}{1}\left(\frac{1}{m'(t)-2}\right)\left(1-\frac{1}{m'(t)-2}\right)^m\Bigg]+$$

$$+\Bigg[\sum_{k=4}^{m'(t)-1}Ph2(k)\Bigg[R_k\left(VoIP\right)\binom{m'(t)-k+1}{1}\left(\frac{1}{m'(t)-k+1}\right)\left(1-\frac{1}{m'(t)-k+1}\right)^l+$$

$$P_{ss}(2)R_k\left(DATA\right)\binom{m'(t)-k+1}{1}\left(\frac{1}{m'(t)-k+1}\right)\left(1-\frac{1}{m'(t)-k+1}\right)^m\Bigg]\Bigg]$$

$$+\left[Ph2\,(m'(t))\left[R_{m'(t)}\left(VoIP\right)+Pss(2)R_{m'(t)}\left(DATA\right)\right]\right]\Big]+\ldots\ldots..$$

$$+\left[Ph1\,(m'(t)-1)\left[R_{m'(t)}\left(VoIP\right)+Pss(2)R_{m'(t)}\left(DATA\right)\right]\right]$$

$$\text{where}\quad \begin{array}{l} l=j-3,\,m=j-4\ \forall\ Pss\,(1),\\ l=j-2,\,m=j-3\ \forall\ \left(1-Pss\,(1)\right)\end{array}$$

(6.71)

6.4.5 Performance Evaluation

This section analyzes the developed Markov models to ascertain the superiority of 2-tier CRN over Basic CRN and also establish the improvement in performance after incorporating spectrum handoff in 2-tier CRN. The key parameters that are used to analyze the performance improvement of 2-tier CRN over Basic CRN include SU transmission, spectrum handoff, blocking and dropping probabilities and overall SU throughput.

Let P_L denote the limiting probability of SU acceptance by available idle channel in CRN and is expressed as follows.

$$P_L = \sum_{\substack{i=0,\\ j=N-i}}^{N-1} P(i, j, 0, 0, 0) \tag{6.72}$$

It is observed from Fig. 6.12 that 2-tier CRN provides higher probability of SU acceptance than Basic CRN and thus reduces the overall blocking probability (denoted by P_B).

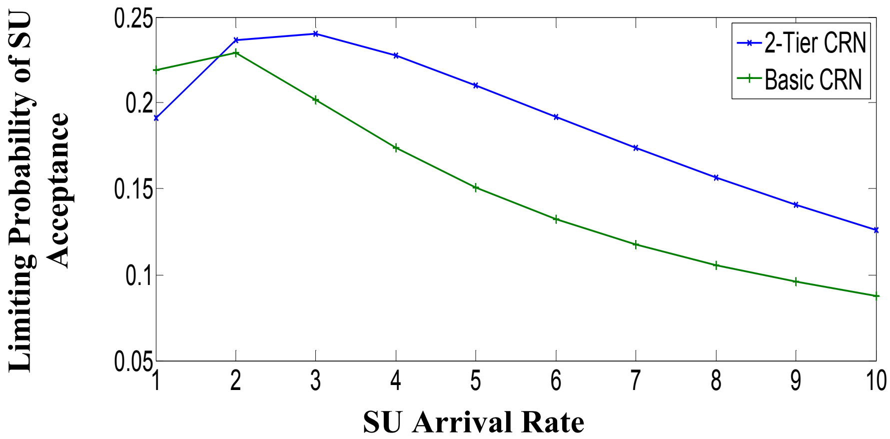


Fig. 6.12 Variation in limiting probability of SU acceptance by CRN with SU arrival rate

Let P_{DR} define the steady state dropping probability that SU transmission is dropped before scheduled transmission interval is over. The expression for P_{DR} is derived from [6.49] and is expressed in (6.73) as per the designed Markov Model.

$$P_{DR} = \frac{\sum P_{drop}}{(1 - P_B)\lambda_s}$$

where $P_{drop} = \sum P(i, j, k, l, m) \ \forall \ (l, m) = (2,2) | (2,0) | (0,2)$ (6.73)

Therefore, spectrum handoff must be performed by SUs to shift to available idle channels on PU arrival to reduce P_{DR}. Let $P_{handoff\_tier1}$ and $P_{handoff\_tier2}$ be the probabilities of spectrum handoff performed by SUtier1 and SUtier2 respectively and are expressed as follows.

$$P_{handoff\_tier1}(i, j) = \frac{\sum P(i, j, 0, 1, 0)}{(1 - P_B)\lambda_s} \tag{6.74}$$

$$P_{handoff\ \_tier\ 2}(i,j)=\frac{\sum P(i,j,0,1,1)}{(1-P_B)\lambda_s} \tag{6.75}$$

The dropping and handoff probabilities for SUs in 2-tier CRN are plotted in Fig. 6.13 and Fig. 6.14 respectively, for two scenarios that correspond to i) Sc. 1: spectrum handoff by SUtier1 only, and ii) Sc. 2: spectrum handoff by SUtier1 and SUtier2.

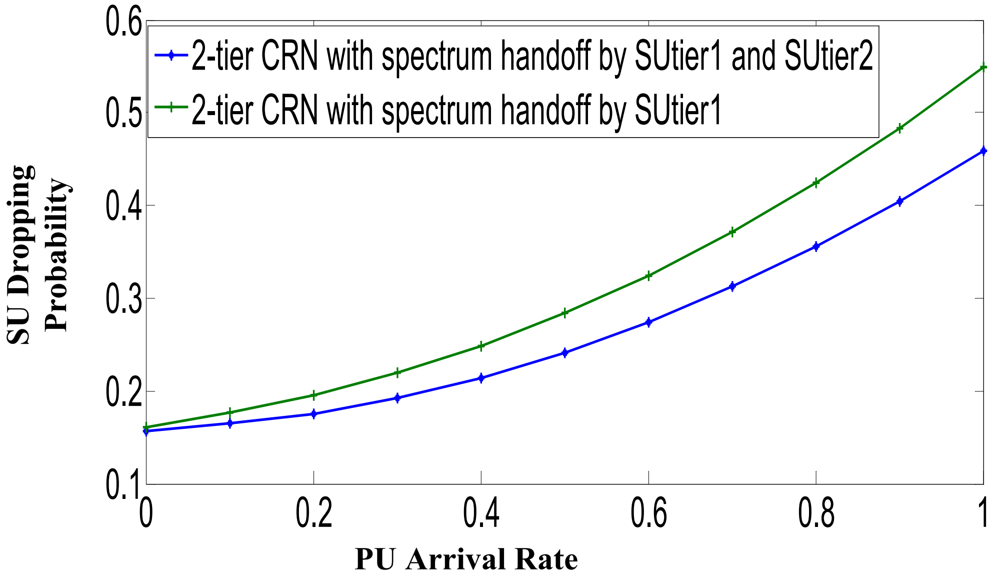


Fig. 6.13 Variation in SU dropping probability in CRN with PU arrival rate

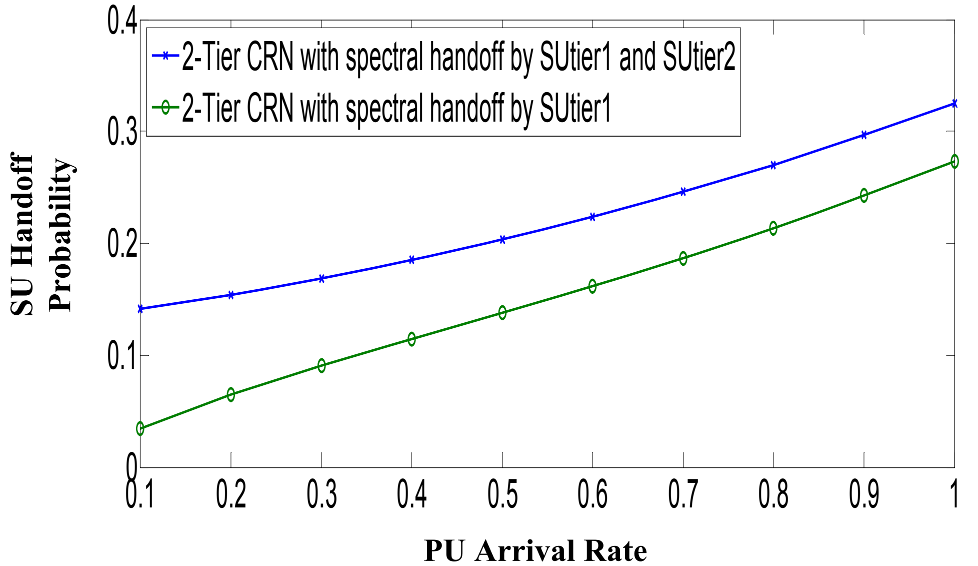


Fig. 6.14 Performance of 2-tier CRN with respect to SU handoff probability for varying PU arrival rate

It is imperative that when SUtier1 performs spectrum handoff, SUtier2 is either dropped or else it must also perform handoff. This situation is clearly reflected in Fig. 6.13 where P_D is less for scenario 2 compared to scenario1.

Therefore, reduction in blocking and dropping probabilities must increase SU throughput in spectrum handoff enabled 2-tier CRN. This is illustrated in Fig. 6.15 that plots the probability of successful transmission by SUs with increase in PU activity. It is observed from the figure that 2-tier CRN with complete spectrum handoff has the highest probability of transmission.

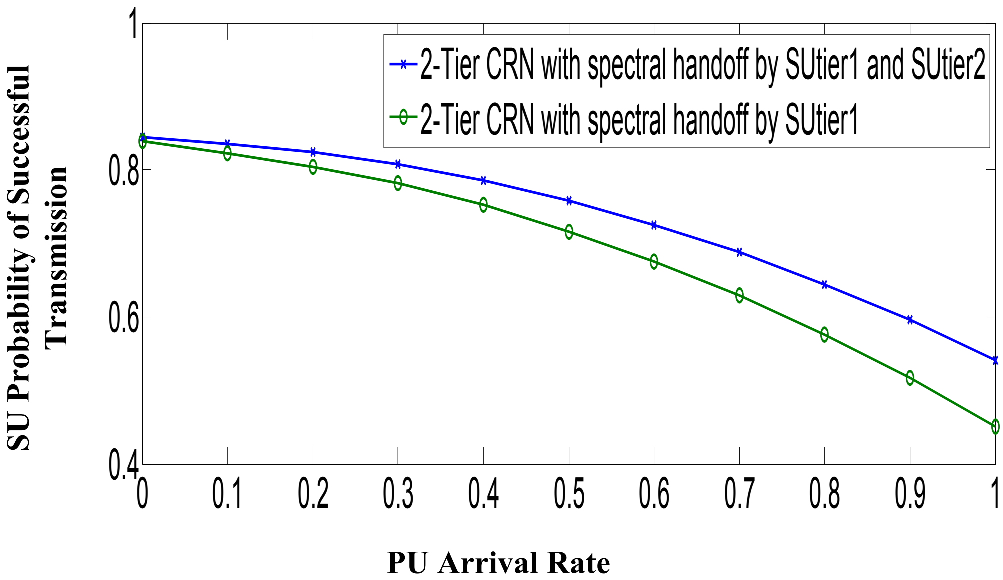


Fig. 6.15 Variation in probability of successful transmission by SU in CRN with PU arrival rate

The normalized SU throughput as obtained from (6.70) is plotted in Fig. 6.16 for increasing probability of PU arrival on the current channel with respect to 2-tier CRN and Basic CRN. As observed from the figure, spectrum handoff enabled 2-tier CRN provides the highest SU throughput compared to the other scenarios. In addition, the CRN implementing spectrum handoff performs better as it records almost 25% enhancement in throughput (for 0.5 PU arrival probability) compared to spectrum handoff disabled CRN. However, as PU activity increases in CRN, the number of idle channels reduces drastically. Under such circumstances, throughput for SUs supporting spectrum handoff decreases and this is reflected in Fig. 6.16.

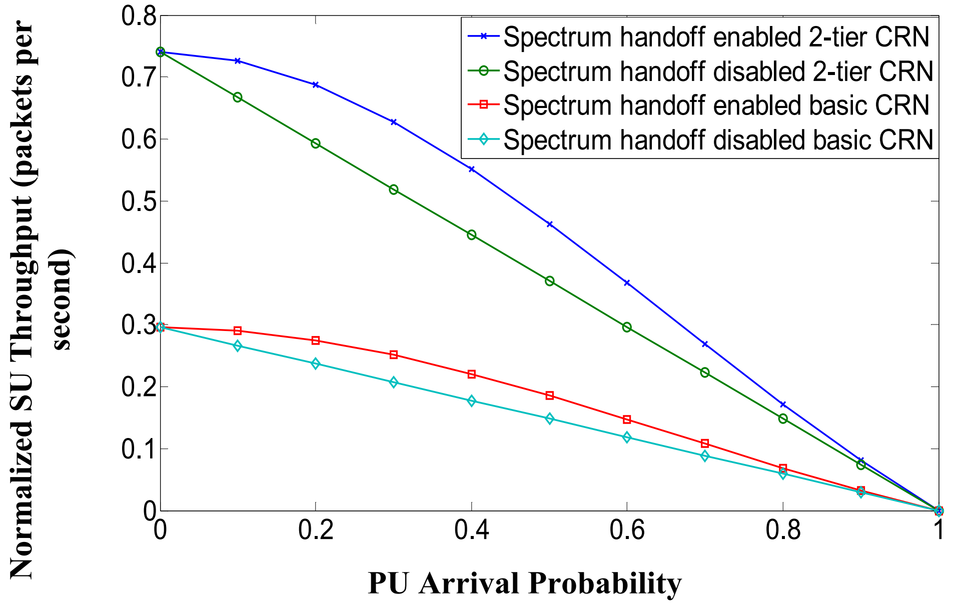


Fig. 6.16 Effect of spectrum handoff on normalized SU throughput with varying PU arrival probability for Basic and 2-tier CRN

Thus, observation from the designed Markov Model in Fig. 6.10 is validated using output derived from the mathematical model in Fig. 6.16. Overall, the Markov Models provide us with a clear picture of how SUtier1 and SUtier2 interact in the 2-tier CRN towards ensuring superior performance in terms of different system probabilities, in comparison to the Basic CRN.

6.5 Developing Simulation Model for 2-Tier CRN

In order to validate the analytical models of VoIP based 2-tier CRN (as designed in Section 6.3) and also observe the aspects of real life-like communication, a simulated model for 2-tier CRN is developed over the established OPNET based model (which has been discussed in Chapter 3).

In the developed VoIP based 2-tier CRN, the Application layer in implements VoIP traffic for SUtier1 and FTP traffic for SUtier2. G.723.1 [6.47] is used as the VoIP codec for SUtier1 that transmits at 33.3 packets per second and supports silence suppression. The on-off interval of SUtier1 is modeled as an exponential distribution with approximate 40% talkspurt time. Data rate of SUtier2 is kept same as that of SUtier1 for comparative performance analysis.

The on-off traffic model for SUtier2 is also exponentially distributed with 50% on time.

Fig. 6.17 is a screenshot obtained during actual simulation of the VoIP based 2-tier CRN for certain duration of secondary transmission interval (t_d) in a particular channel. It is observed that there are sufficient idle time periods during VoIP transmission by SUtier1, that are accessed by SUtier2. It has been deduced from Proposition 1 that the minimum increase in SU Sum Goodput for 2-tier CRN is twice the SU Sum Goodput in Basic CRN. However, in a real scenario, SUtier2 performs multiple transmissions in an idle channel, leading to 4 times increase in throughput for 2-tier CRN with respect to Basic CRN as seen in Fig. 6. 18. Intuitively, the average link utilization percentage in 2-tier CRN rises by 40% in Fig. 6.19 compared to Basic CRN for identical SU transmission rates.

It is also observed from Fig. 6.20 that 2-tier CRN serves 138 SUtier1 and 90 SUtier2 simultaneously. This is because SUtier1 allows SUtier2 to transmit only when it has enough idle time slots, as governed by the value of P_{ss} in that channel. Hence, Fig. 6.20 validates the range of SUtier2 that can be admitted in 2-tier CRN as stated in Proposition 2.

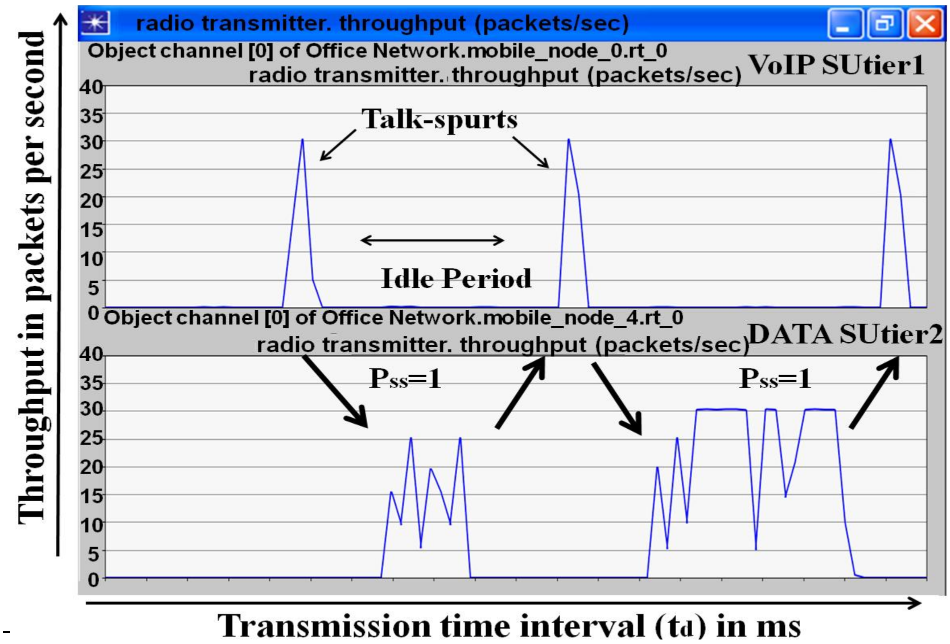


Fig. 6.17 Screenshot of the ongoing simulation in OPNET Modeler 16.0.A. depicting channel access by SUtier2 in the idle periods of SUtier1

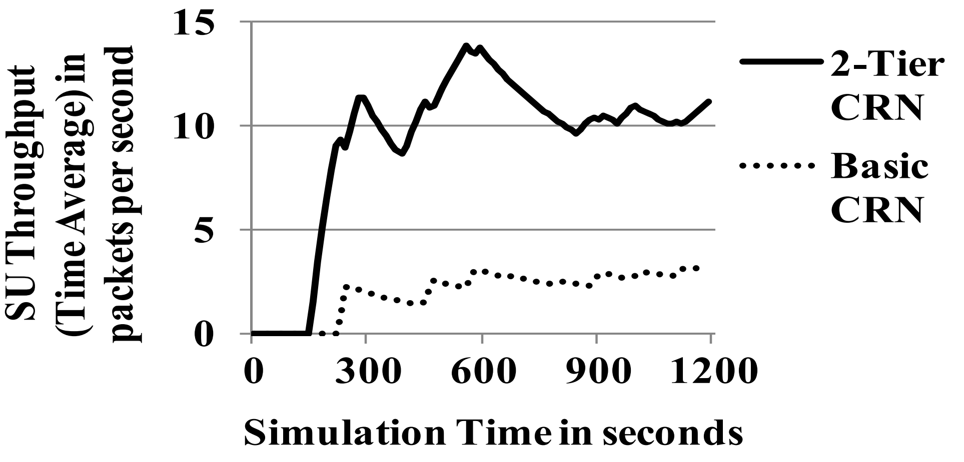


Fig. 6.18 Time Average SU Throughput for Basic and 2-tier CRN scenarios

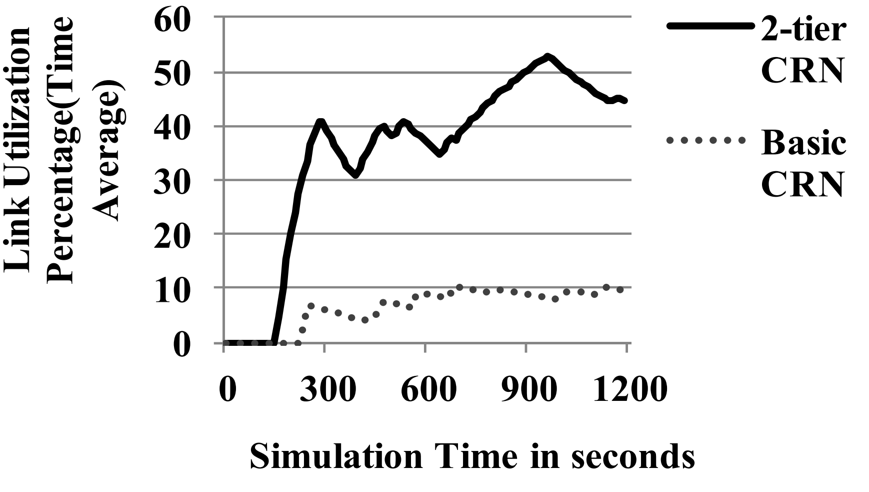


Fig. 6.19 Time Average Link Utilization Percentage for both Basic and 2-tier CRN scenarios

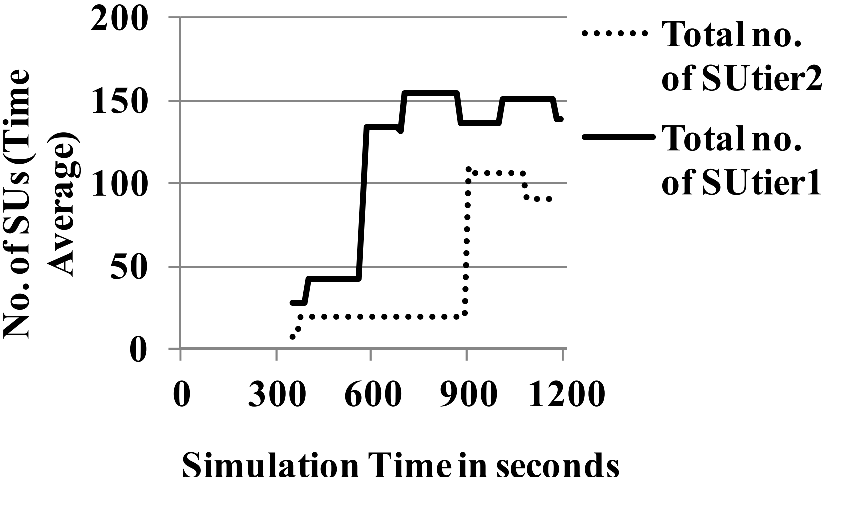


Fig. 6.20 Total No. of SUtier1 and SUtier2 (Time Average) for both Basic and 2-tier CRN scenarios

Thus, real-time simulation output in this section validates the findings of the mathematical models in Section 6.3.

6.6 Formulation of Implementation Techniques for Practical Deployment

As the success of any technological advancement relies on efficient implementation design policies, this section aims to design novel practical implementation strategies for VoIP based 2-tier CRN. In this regard, the primary objective is to devise implementation algorithms for allowing transmission by SUs in both tiers of CRN. Two algorithms, namely *Simple_msg* and *Periodic_msg* are proposed along with mathematical modeling and performance evaluation to determine the conditions under which each algorithm can be applied successfully.

6.6.1 Design Considerations

This section deals with the design considerations for VoIP based 2-tier CRN. Firstly, 2-tier CRN architecture must implement message passing mechanism as the primary means of interaction among the SUs. As SUtier1 and SUtier2 host VoIP and DATA applications respectively, they have different message formats. Hence, their messages must be decoded by the corresponding VoIP and DATA servers to enable SUs to take appropriate action as per the contents of the message.

It is considered that SUs implement optimal selection of cognitive cycle timing parameters (sensing time, transmission time) to ensure PU arrival at end of transmission time slot (t_d). Appropriate queuing mechanism must be in place for VoIP traffic in case of delay incurred during channel release from SUtier2 to SUtier1. As algorithms discussed in this section propose channel release prior to onset of new talkspurt, there is little probability of excess VoIP packets waiting at queue. Hence, discussion of queuing theory in this aspect is insignificant and is not dealt with in this section. Moreover, suitable MAC protocols must be implemented to select SUtier2 from a set of SUs for transmission in the second tier of CRN.

Table 6.4 Message Definitions for implementing the proposed algorithms

Message	Source	Destination	Purpose
INVITE	SUtier1	SUtier2	Request for channel access
INVITE_ACK	SUtier2	SUtier1	Channel access confirmed
CHECK	SUtier2	SUtier1	Request renewal of transmission
RELEASE	SUtier1	SUtier2	Request release of channel
RELEASE_ ACK	SUtier2	SUtier1	Channel release confirmed

6.6.2 Proposed Algorithms for Implementing VoIP based 2-Tier CRN

Two algorithms are designed in this section to allow SUtier2 transmission in VoIP based 2-tier CRN. The messages to be used by both the algorithms are defined in Table 6.4.

(i) Simple Message Passing Algorithm – Simple_msg

Simple_msg algorithm aims to maximize system throughout by minimizing time of interaction between SUtier1 and SUtier2. According to this algorithm, once silence suppression mode is activated, SUtier1 sends INVITE message to SUtier2 to allow access to the channel. SUtier2 acknowledges it by sending INVITE_ACK message and initiates transmission in that channel. It continues transmission unless any message arrives from SUtier1. SUtier1 sends RELEASE message to SUtier2 for channel release at time t_1 prior to either the onset of a talkspurt or end of t_d. t_1 time is taken as time consumed in a INVITE-INVITE_ACK message exchange. The arrival of new talkspurt for SUtier1 is estimated using previously collected statistical data regarding onset of talkspurt as obtained from VAD (Voice Activity Detector) or using estimated talkspurt value. Finally, SUtier2 receives RELEASE message from SUtier1, suspends transmission and sends RELEASE_ACK as acknowledgement. The pseudo code is described as follows.

/*Pseudo Code for Simple_msg Algorithm*/	
Step 1:	For *every silence suppression activated*{
Step 2:	*SUtier1 sends INVITE to SUtier2.*
Step 3:	*SUtier2 replies with INVITE_ACK.*
Step 4:	*SUtier2 initiates transmission.*
Step 5:	do{
Step 6:	*SUtier2 continues data transmission.*
Step 7:	If *(New talkspurt is activated for SUtier1 after t_1 time*|| *t_d is over after t_1 time)* {
Step 8:	*SUtier1 sends RELEASE to SUtier2.* }
Step 9:	}while !*(SUtier1 sends RELEASE)*
Step 10:	*SUtier2 suspends current transmission.*
Step 11:	*SUtier2 sends RELEASE_ACK to SUtier1.*

(ii) Periodic Message Passing Algorithm – Periodic_msg

The drawback of *Simple_msg* algorithm is that it introduces interference among SUs. This results in packet loss and hence, must be reduced to ensure sufficient QoS of VoIP call for SUtier1. *Periodic_msg* algorithm aims to minimize this time of interference by periodically checking for either the end of t_d or onset of a new talkspurt for SUtier1.

Periodic_msg algorithm implements the entire message-exchange procedure as used in *Simple_msg*. In addition, it incorporates periodic checking mechanism where SUtier2 periodically suspends transmission and sends CHECK message to SUtier1. SUtier1 checks for arrival of new talkspurt or end of t_d. If either of the above conditions is true, SUtier1 replies with RELEASE message. Otherwise, it sends INVITE message. SUtier2 resumes transmission only after receiving INVITE message and acknowledging with INVITE_ACK message. The time interval before which SUtier2 temporarily suspends its transmission and performs periodic checking is derived in Section 6.4. The pseudo code for *Periodic_msg* is given as follows.

*/*Pseudo Code for Periodic_msg Algorithm*/*	
Step 1:	For *every silence suppression activated* {
Step 2:	*SUtier1 sends INVITE to SUtier2.*
Step 3:	*SUtier2 replies with INVITE_ACK.*
Step 4:	*SUtier2 initiates transmission.*
Step 5:	do {
Step 6:	*Periodic Interval is reset to default value.*
Step 7:	do{
Step 8:	*SUtier2 continues data transmission.*
Step 9:	*Periodic Interval is decremented.*
Step 10:	} while !*(Periodic Interval is over)*
Step 11:	*SUtier2 sends CHECK to SUtier1.*
Step 12:	If *(New talkspurt is activated for SUtier1 after t_1 time|| t_d is over after t_1 time)* {
Step 13:	*SUtier1 sends RELEASE to SUtier2.* }
Step 14:	Else{
Step 15:	*SUtier1 sends INVITE to SUtier1.* }
Step 16:	} while *(SUtier1 sends INVITE)*
Step 17:	*SUtier2 suspends current transmission.*
Step 18:	*SUtier2 sends RELEASE_ACK to SUtier1.*}

Both the algorithms are illustrated in Fig. 6.21.

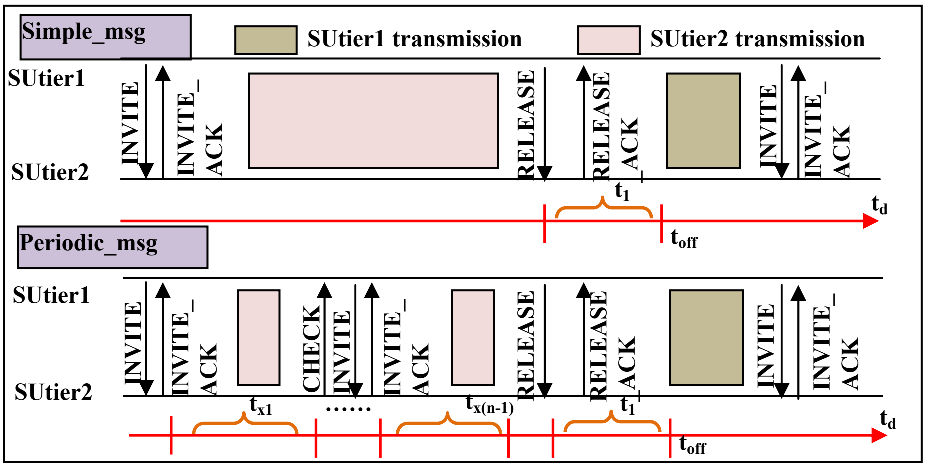


Fig. 6.21 Schematic Representation of *Simple_msg* and *Periodic_msg* Algorithms

6.6.3 Evaluation of Performance Parameters for the Designed Implementation Algorithms

This section deals with deriving mathematical expressions for total SUtier2 transmission time, time incurred in message passing, overall interference time and channel utilization percentage for *Simple_msg* and *Periodic_msg* algorithm. It is quite obvious that increasing the interaction among the SUs in both the tiers in terms of message-passing will mitigate the issues arising out of interference with the randomly arriving PUs. However, more messages imply significant delay overhead, resulting in an effective loss in communication duration. In order to solve this problem, this section analytically derives the related system performance parameters and analyzes them with respect to both *Simple_msg* and *Periodic_msg* algorithms. The significance of the findings in this section is truly realized through simulation studies in the next section that not only verify and validate the analytical observations but also determines the conditions where each algorithm can be efficiently applied.

(i) Mathematical Background

The primary time components involved in message passing in the designed algorithms are defined as follows.

1. d_0= time consumed in INVITE-INVITE_ACK message + time required by SUtier2 to initiate transmission

2. d_1= time taken by SUtier1 to send RELEASE message to SUtier2.

3. d_2= time consumed in decoding messages and taking appropriate actions. In case of receipt of RELEASE message, d_2 is the time taken by SUtier2 to stop its transmission and send RELEASE_ACK to SUtier1. For CHECK-INVITE message exchange, d_2 is the time taken by SUtier2 to send INVITE_ACK and resume transmission. It is considered that the average time consumed by SUtier2 is same for both initiating transmission and suspending an ongoing transmission.

4. d_3= time taken by SUtier2 to suspend transmission and send CHECK message to SUtier1+time consumed by SUtier1 to send INVITE/RELEASE message.

As VoIP transmission occurs in the form of talkspurts, an on-off model is considered in SUtier1 with α^{-1} and β^{-1} as on and off periods respectively, that are exponentially distributed [6.36]. Therefore, the probability with which SUtier1 is idle is given by,

$$P_{idle} = \frac{\beta^{-1}}{\alpha^{-1} + \beta^{-1}} \tag{6.76}$$

Let T_{idle} be the total time for which SUtier1 is in silent suppression mode in t_d transmission time slot and is given by,

$$T_{idle} = t_d \times P_{idle} \tag{6.77}$$

(ii) ***<u>Mathematical Formulation of Simple_msg Algorithm</u>***

Let SUtier2 transmits for t_x time in the idle time period in a single on-off interval (denoted by t_{off}).

Hence, t_{off} = transmission initiation time + SUtier2 transmission time + channel release time

$$\Rightarrow t_{off} = d_0 + t_x + d_1 + d_2 \Rightarrow t_x = t_{off} - d_{Simple\_msg} \tag{6.78}$$

where d_{Simple_msg} is the time incurred in message passing and is expressed as,

$$d_{Simple\_msg} = d_0 + d_1 + d_2 \tag{6.79}$$

Considering g as the total number of off periods for SUtier1 in 1 t_d time slot, (6.79) can be generalized as

$$D_{Simple\_msg} = \sum_{i=1}^{g} (d_0 + d_1 + d_2) \tag{6.80}$$

where D_{Simple_msg} is the time taken in message passing in 1 t_d interval. There are two scenarios where interference occurs during channel release by SUtier2. Firstly, interference occurs between SUtier1 and SUtier2 during channel release at the onset of a talkpurt. The second scenario witnesses interference between SUtier2 and PU when PU arrives at end of t_d. Let $T\_inf_{Simple\_msg}$ be total interference time and is given by,

$$T\_inf_{Simple\_msg} = \sum_{i=1}^{g} (d_1 + d_2 - t_1) \tag{6.81}$$

Therefore, total time of successful transmission by SUtier2 in 1 t_d interval is derived from (6.77), (6.80) and (6.81) as,

$$T_{Simple\_msg} = T_{idle} - \left(D_{Simple\_msg} + T\_inf_{Simple\_msg}\right)$$

$$= \left(t_d \times P_{idle}\right) - \left(\sum_{i=1}^{g}(d_0 + d_1 + d_2) + \sum_{i=1}^{g}(d_1 + d_2 - t_1)\right) \tag{6.82}$$

Expressing (6.82) as percentage of the total silence period,

$$Utilization\_Percent_{Simple\_msg} = \frac{T_{Simple\_msg}}{T_{idle}} \times 100 \tag{6.83}$$

(iii) <u>Mathematical Formulation of Periodic_msg Algorithm</u>

Let there be "n" instances where message exchange occurs between SUtier1 and SUtier2 in *Periodic_msg* algorithm. This implies that there are (n-2) such instances where SUtier2 suspends transmission and checks for availability of channel. SUtier2 therefore, transmits for (n-1) number of instances. Let Pr_i denote the probability that RELEASE message has not been sent by SUtier1 in the ith instance, thereby allowing SUtier2 to transmit. Let t_{xi} be the SUtier2 transmission time in the ith instance. Therefore, t_{off} as defined in (6.78) is modified as follows.

$$t_{off} = d_0 + t_{x1} + \{Pr_2 \times (d_3 + d_2 + t_{x2}) + (1 - Pr_2) \times (d_3 + d_2)\} +$$

$$\cdots + \left[\prod_{i=1}^{n-2} Pr_i \times \{\{Pr_{n-1} \times (d_3 + d_2 + t_{x(n-1)})\} + \{(1 - Pr_{n-1}) \times (d_3 + d_2)\}\}\right]$$

$$+ \prod_{i=1}^{n-1} Pr_i \times (d_1 + d_2) \quad where \quad Pr_1 = 1 \tag{6.84}$$

Considering g as the total number of off periods for SUtier1 in 1 t_d time slot, (6.84) can be used to derive the total time incurred in message passing and is expressed as,

$$D_{Periodic\_msg} = \sum_{l=1}^{g} \left\{ d_0 + (d_1 + d_2) \prod_{k=1}^{n(l)-1} Pr_k + (d_2 + d_3) \left\{ 1 + \sum_{v=2}^{n(l)-2} \left(\prod_{k=1}^{v} Pr_k \right) \right\} \right\}$$

$$where \qquad n = \sum_{l=1}^{g} n(l) \tag{6.85}$$

Interference for *Periodic_msg* algorithm is recorded when SUtier2 fails to identify the channel status during periodic checks. This occurs when SUtier1 sends RELEASE message after SUtier2 has resumed its transmission following successful CHECK-INVITE-INVITE_ACK message exchange. The total time of interference in t_d time slot is given by,

$$T\_inf_{Periodic\_msg} = \sum_{l=1}^{g} \left\{ (d_1 + d_2 - t_1) \prod_{k=1}^{n(l)-1} Pr_k \right\} \tag{6.86}$$

Hence, total transmission time for SUtier2 in 1 t_d interval is derived from (6.77), (6.85) and (6.86) and is expressed in (6.87).

$$T_{Periodic\_msg} = \frac{(td \times Pidle) - \left[\sum_{l=1}^{g} \left\{ d0 + (d1 + d2) \prod_{k=1}^{n(l)-1} Prk + (d2 + d3) \left\{ 1 + \sum_{v=2}^{n(l)-2} \left(\prod_{k=1}^{v} Prk \right) \right\} \right\} \right]}{1 + \sum_{j=2}^{n-1} \left(\prod_{k=1}^{j} Prk \right)}$$

$$- \sum_{l=1}^{g} \left\{ (d1 + d2 - t1) \prod_{k=1}^{n(l)-1} Prk \right\} \tag{6.87}$$

The percentage of silence period utilized by SUtier2 is given by,

$$Utilization\_Percent_{Periodic\_msg} = \frac{T_{Periodic\_msg}}{T_{idle}} \times 100 \qquad (6.88)$$

It is interesting to note here that the expression for total transmission time in *Simple_msg* algorithm, as given by (6.82) is pretty straightforward and incorporates all the delay components due to message exchanges that occur only at the beginning and the end of each SUtier2 transmission. On the other hand, the same expression for *Periodic_msg* algorithm as in (6.87) is influenced by several probability terms that arise due to periodic message exchanges. Thus, *Periodic_msg* is more complicated in nature but can prove effective especially under highly random PU activities. The same is realized through simulation observations in the next section.

6.6.4 Comparative Performance Evaluation of the Proposed Algorithms

The proposed algorithms are evaluated for performance efficiency. Initially, total number of talkspurts in t_d is kept constant. It is observed from Fig. 6.22 that SUtier2 transmission time decreases with rise in talkspurt percentage for SUtier1. As noticed from Fig. 6.22, *Simple_msg* provides better utilization of t_{off} period by SUtier2 than *Periodic_msg* as validated from (6.81) and (6.86). Moreover, 2-tier CRN records 77% increase in utilization of t_d (98.5% for *Simple_msg* and 98.34% for *Periodic_msg*) compared to Basic 1-tier CRN (21.42 %).

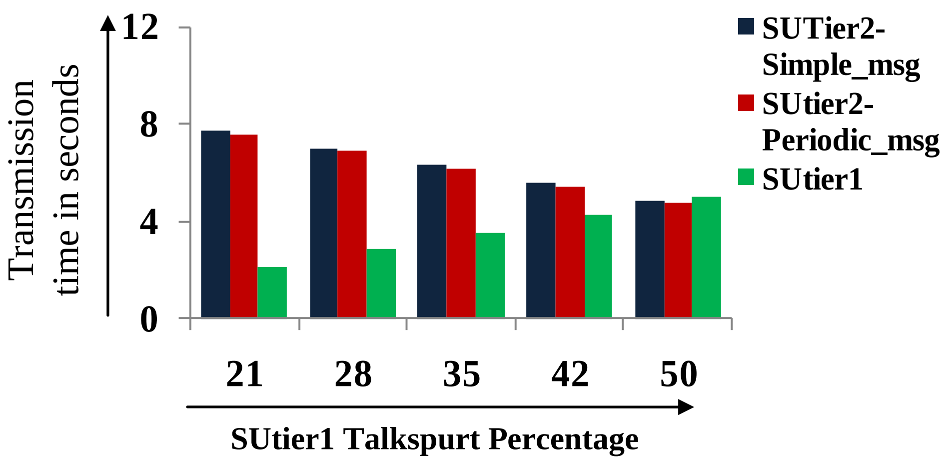


Fig. 6.22 Variation in SU Transmission Time with SUtier1 talkspurt percentage

The algorithms are further evaluated with respect to silence detectors used in modern communication [6.9] as described in Table 5.2 in Chapter 5. It is inferred from Fig. 6.23 that the total time consumed in message passing is much less in *Simple_msg* compared to *Periodic_msg*, as confirmed from (6.80) and (6.85). An important observation from Fig. 6.23 is that not only the talkspurt percentage but also the total number of talkspurts (h) in t_d time frame affects the percentage of channel utilization by SUtier2. It is shown in Fig. 6.23 that as h increases, percentage loss in channel utilization for *Periodic_msg* rises (11.9% for h=13 compared to 2.6% for h=3).

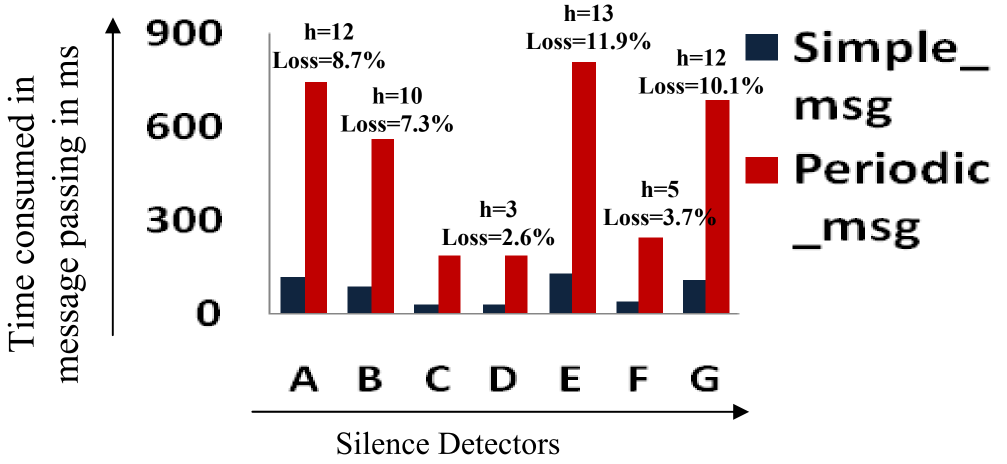


Fig. 6.23 Effect of different silence detectors as mentioned inTable 5.2 on transmission time wasted due to message exchange

Moreover, *Simple_msg* records increased interference time (almost thrice) than *Periodic_msg* as illustrated by the shaded region in Fig. 6.24 and validated from (6.81) and (6.86). As interference with SUtier1 degrades QoS of VoIP calls, it is concluded that *Periodic_msg* is more suitable than *Simple_msg* for high SUtier1 activity. *Simple_msg* algorithm must be implemented for low SUtier1 activity as it ensures maximum utilization of the silence periods.

Performance of *Periodic_msg* is highly dependent on selection of suitable SUtier2 transmission time interval before sending CHECK message to SUtier1. As observed from Fig. 6.25 and Fig. 6.26, with increase in number of intervals for periodic checks, time consumed in message passing increases while interference time decreases.

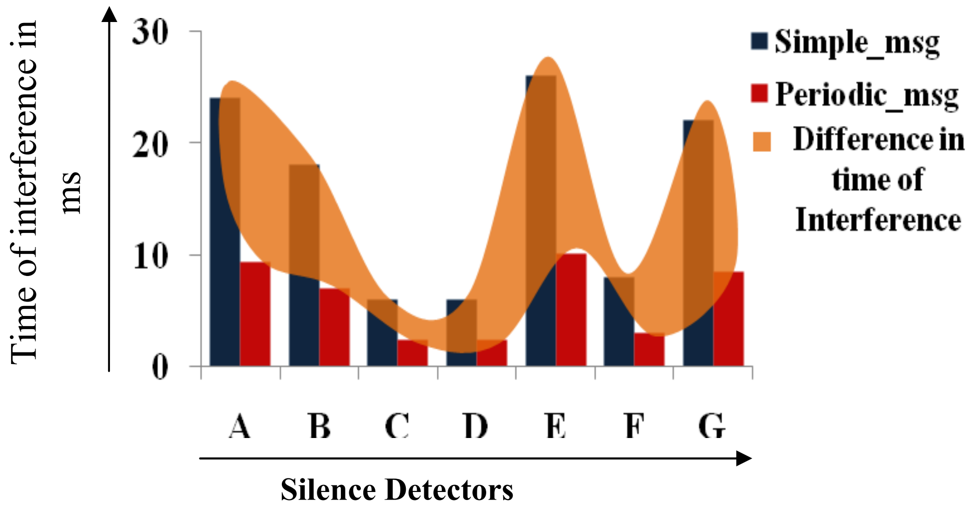


Fig. 6.24 Variation in total interference time for different silence detectors as described in Table 5.2

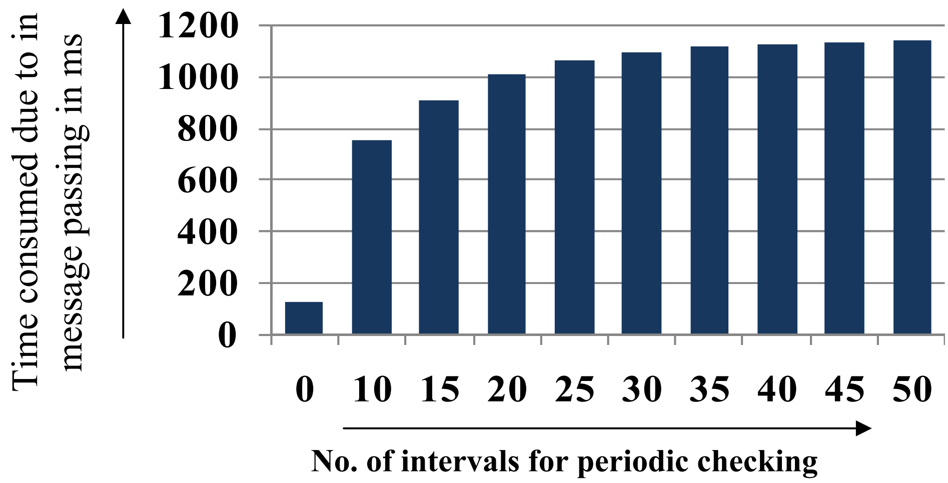


Fig. 6.25 Effect of no. of intervals during periodic checking process in *Periodic_msg* on the total Time consumed in message passing

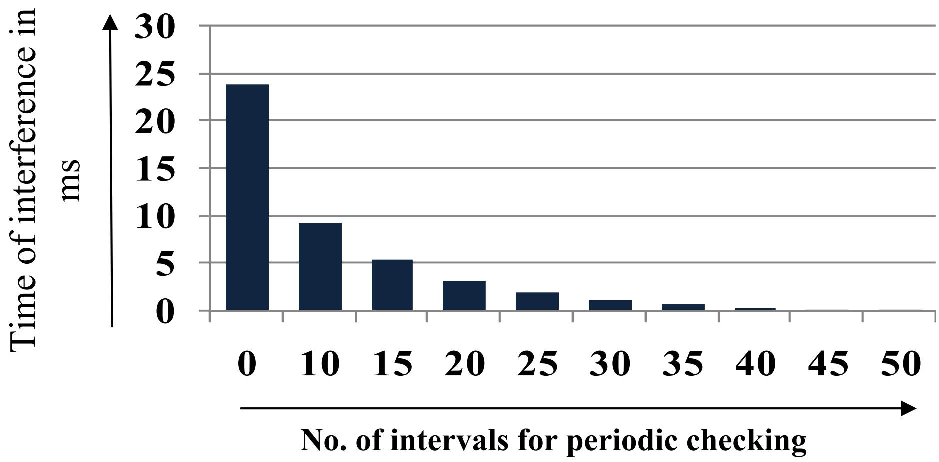


Fig. 6.26 Effect of no. of intervals during periodic checking process in *Periodic_msg* on the interference time in *Periodic_msg*

In order to achieve tradeoff between time lost in message exchange and interference time, total number of CHECK instances as denoted by m_{CHECK} must be equal to the number of estimated talkspurts. Hence, SUtier2 transmission time in between successive CHECK messages is given by,

$$t\_interval_{Periodic\_msg} = \frac{t_{off} - \{m_{CHECK} \times (d_3 + d_2)\}}{m_{CHECK}} \quad (6.89)$$

Thereafter, the algorithms are implemented in the simulation model as discussed in Section 6.5. There are two reasons for this analysis. Firstly, the real life-like performance of these algorithms must be observed in a comprehensive simulation model such as OPNET Modeler. Secondly, the mathematical formulation of these algorithms followed by analysis in the previous sections requires validation to establish their credibility. Hence, the designed model in Section 6.5 is considered for performance evaluation in this section.

It is observed from Fig. 6.27 that *Simple_msg* provides higher throughput than *Periodic_msg* and both these algorithms ensure better spectrum utilization than the Basic CRN. Fig. 6.28 confirms that the time wasted in message passing is much higher in *Periodic_msg* than *Simple_msg* (where it is negligible). It is also noticed from Fig. 6.29 that *Periodic_msg* ensures more number of SU transmission instances with negligible interference than *Simple_msg*.

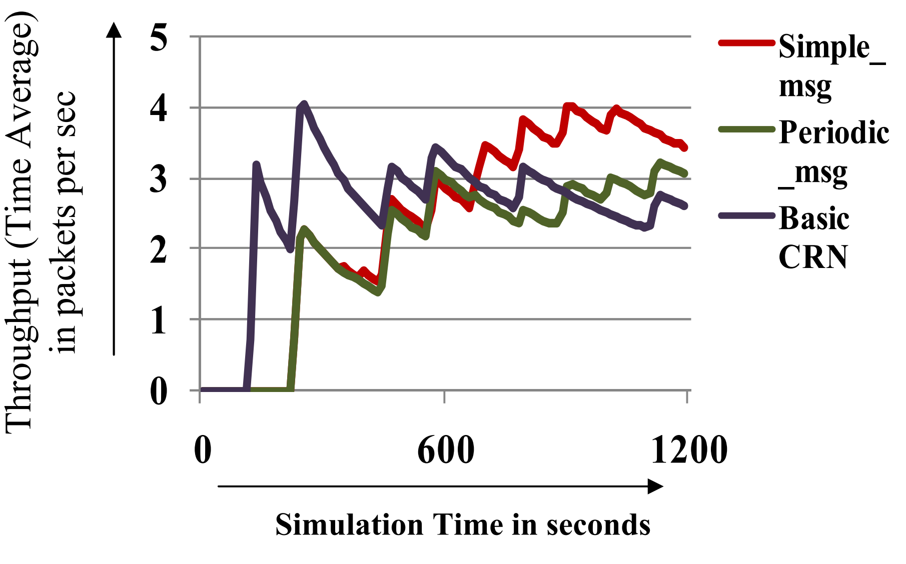


Fig. 6.27 Performance Analysis in OPNET Modeler 16.0.A. with respect to Throughput under different CRN scenarios

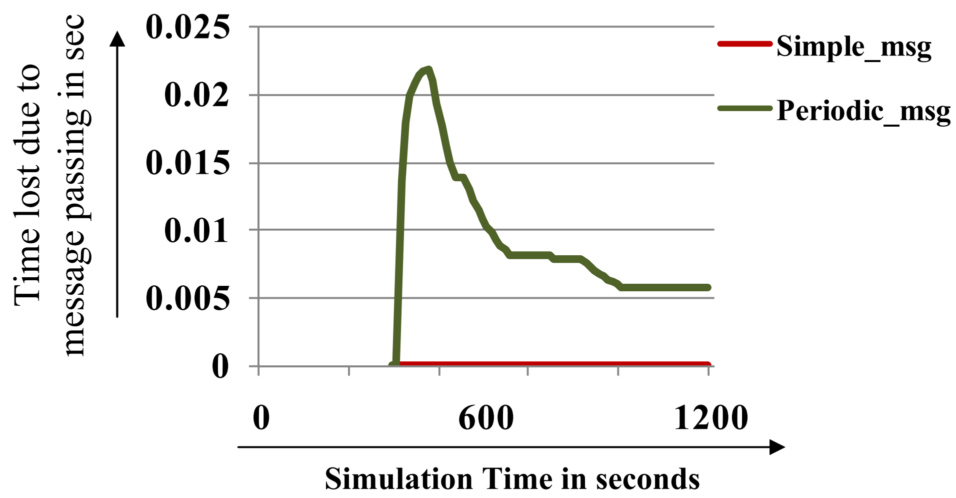


Fig. 6.28 Time Consumed in message passing as analyzed in OPNET Modeler 16.0.A. with respect to different CRN scenarios

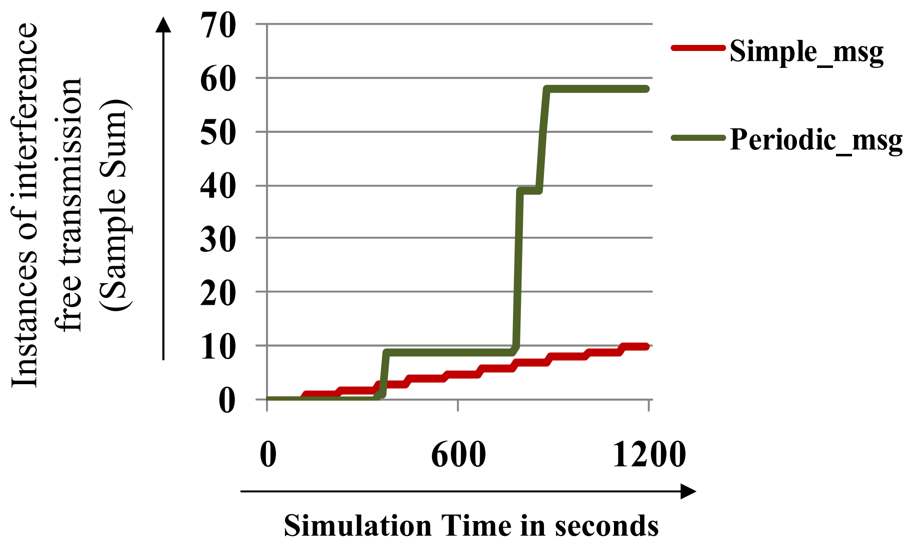


Fig. 6.29 Instances of successful transmission with negligible interference in OPNET Modeler 16.0.A. under different CRN scenarios

Overall, it is inferred that Simple_msg is more effective under the conditions of low and intermittent PU activity as it minimizes the delay incurred due to exchange of messages. Periodic_msg, on the other hand, is more suitable for scenarios having high PU activity, where it minimizes the risk of interference in transmissions.

Thus, the output from the simulation model validates the inferences drawn from mathematical models regarding applicability of the designed algorithms towards building a robust CRN having two tiers of SUs.

6.7 Test-Bed Implementation of VoIP Based 2-Tier CRN

This section highlights the practical significance of VoIP based 2-tier CRN by implementing it in a practical test-bed. As a case-study, the utility of 2-tier CRN is studied with respect to disaster management system.

6.7.1 Description of the Test-bed

The test-bed comprises of WARP (Wireless Open-Access Research Platform) v3 kits [6.50], Wi-Fi enabled computers and an operating Wi-Fi network, as shown in Fig. 6.30. WARP is a scalable wireless platform that integrates programmable hardware (high-performance FPGA with flexible RF peripherals) with open-source repository of reference designs, thereby enabling us to implement our proposed network.

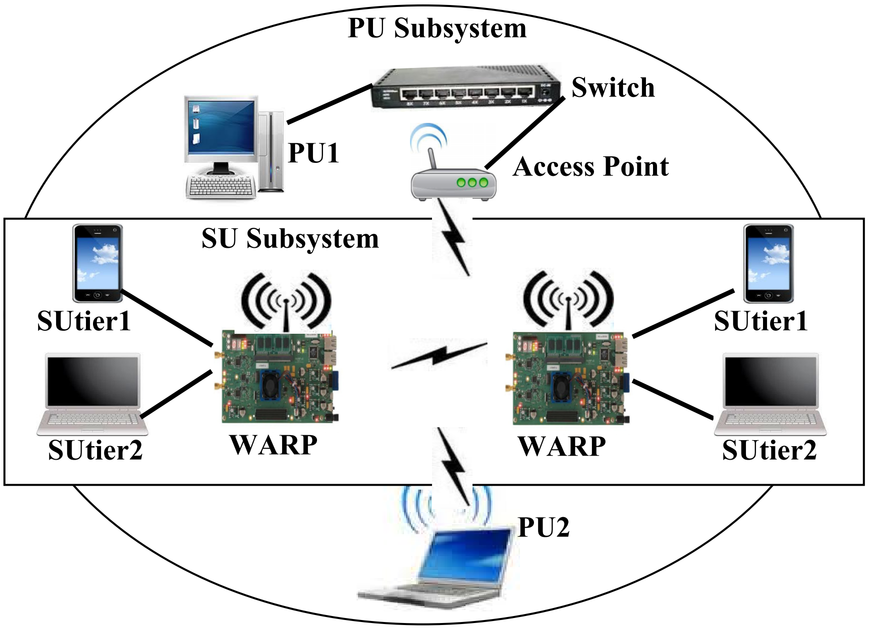


Fig. 6.30 Test-bed model of VoIP based 2-tier CRN

WARP v3 kits are comprised of Xilinx Virtex-6 LX240T FPGA and are used to implement cognitive radio. They also assume the role of spectrum controller. SJPhone [6.51] is the softphone used by VoIP SUtier1. NetScanTools Pro [6.52] generates data traffic and represents SUtier2. SUtier1 and SUtier2 are connected to WARP via Ethernet links. Further, ManageEngine

VQManager [6.53] is used as the monitoring tool to analyze QoS metrics and network throughput.

As WARP operates in the unlicensed ISM (2.4GHz) band, an external Wi-Fi network is developed with HP ProCurve Switch and MSM310 Access Points [6.54] to serve as the primary network as depicted in Fig. 6.30. PUs comprise of a Wi-Fi enabled laptop and a standard computer connected via Ethernet to the switch. A detailed overview of all the hardware and software components is provided in Chapter 8 that solely deals with test-bed model design and implementation.

6.7.2 A Practical Case-Study in Test-bed: Implementing VoIP based 2-tier CRN in Disaster Management System

The Red Cross [6.55] defines disaster management as organization and management of resources and responsibilities for dealing with all humanitarian aspects of emergencies. This section considers such an emergency situation where primary communication is partially disrupted due to natural disaster. The objectives are to i) restore secondary communication among officials for better cooperation, and ii) transfer critical information about casualties, aid services, etc. to the Office.

VoIP based 2-tier CRN is deployed to meet these two objectives. While SUtier1 performs VoIP transmission among the officials, SUtier2 transmits emergency data in the same channel. Channel 9 (2.452 GHz) is used for both PU and SU systems. The PUs in the proposed scenario can still generate traffic with very low probability. Hence, the time for which PU is absent is kept sufficiently long enough for SUs to transmit. Accordingly, sensing time of SUtier1 is kept at 0.03 ms to enable fast setup of VoIP calls.

SJphone uses H.323 [6.46] for VoIP call management as it supports peer-to-peer communication without centralized controlling entity, making it suitable in this system. Advanced features like H.323 tunneling, early H.323 and H.323 fast start are also implemented. Moreover, SJphone uses iLBC codec [6.47] with 30 ms sample interval for SUtier1 transmission and enables silence suppression for SUtier2 to transmit.

WARP being the spectrum controller node confirms the idle period of SUtier1 and allows SUtier2 to transmit. This process undertakes approximately 60 μs (two system clock cycles of 30 μs). SUtier2 finally transmits at the mean rate of 510 kbps (165 packets per second). After every transmission, SUtier2 polls the WARP for channel status and suspends its activity if SUtier1 becomes active. QPSK (Quadrature Phase Shift Keying) modulation is implemented during physical transmission. Carrier sensing threshold is set to -65dBm Receiver power. The minimum duration required to assert packet detection is fixed at 24 system clock cycles.

Fig. 6.31 shows different stages of bandwidth consumption in 2-tier CRN. It is seen in Fig. 6.31 (i) that H.323 consumes high bandwidth during call initiation, which is slowly replaced with VoIP traffic in Fig. 6.31 (ii) as the conversation starts. Introducing SUtier2 in the system further allows 60% of the bandwidth to be utilized by data traffic as shown in Fig. 6.31 (iii). This exhibits better utilization of idle time periods by 2-tier CRN, leading to increased spectrum usage.

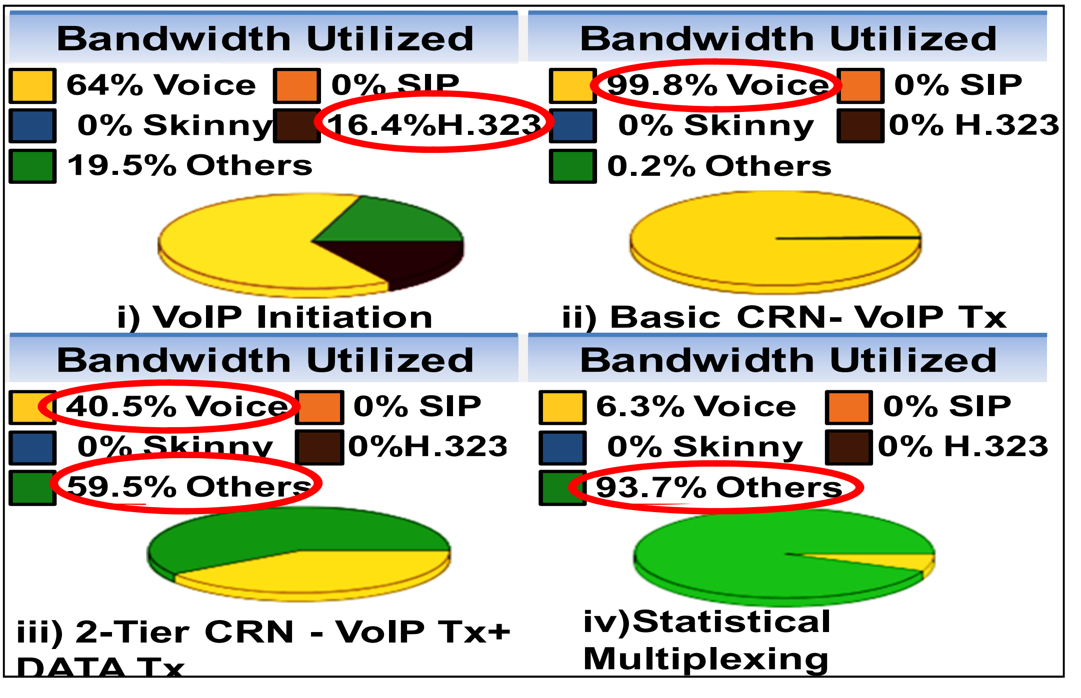
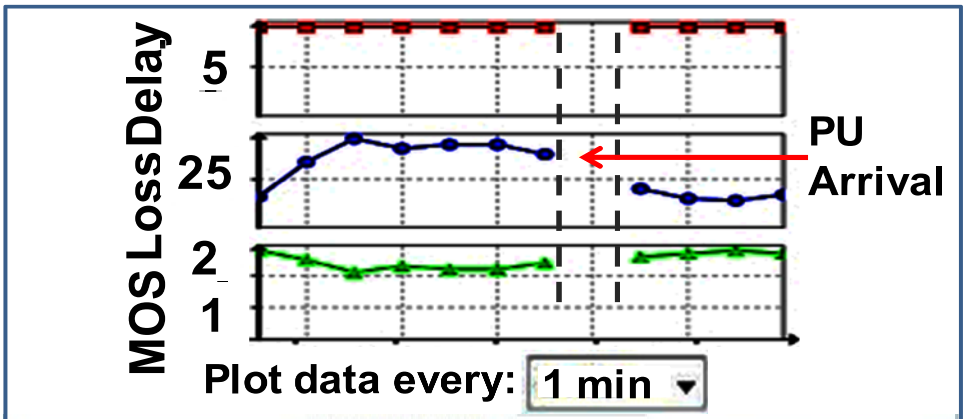


Fig. 6.31 Bandwidth Consumption in different scenarios where 'Others' denote data traffic in the network

It is already discussed in Section 6.2 that 2-tier CRN in this work does not implement statistical multiplexer owing to several practical difficulties. In order to demonstrate these problems, a separate experiment is conducted under identical conditions where a statistical multiplexer controls transmissions over the idle channel. As SUtier1 does not have primary control over this channel, it cannot reclaim the channel back during SUtier2 transmission. Rather, SUtier1 continuously senses the channel to be busy and defers its transmission, severely degrading the VoIP call in this process. SUtier2 continues transmitting and consumes nearly the entire bandwidth as shown in Fig. 6.31 (iv). The proposed 2-tier CRN, on the contrary, experiences good quality VoIP call with mean MOS greater than 3.5 and delay and jitter within 9 ms, as confirmed by Fig. 6.32. However, with PU arrival, packet loss rises and MOS degrades, before SUtier1 suspends its activity on that channel, as indicated by the graph in Fig. 6.32.



	Min	Max	Avg
Delay (ms)	8	9	9
Jitter (ms)	9	9	9

Fig. 6.32 VoIP QoS metrics in VQManager

In order to study the energy efficiency [6.7] of 2-tier CRN, measurements are taken by connecting ammeter in series with WARP node's 12V power supply. It is seen that WARP draws 1.4A and 1.35A current when SU is transmitting and receiving respectively. SU power consumption is 16.8W (12V*1.4A) on transmission and 16.2W (12V*1.35A) on reception. It is to be noted that the power specifications of WARP are different from commercial network cards as FPGA board is the primary source of power consumption in WARP unlike commercial wireless cards where a transceiver draws the maximum power.

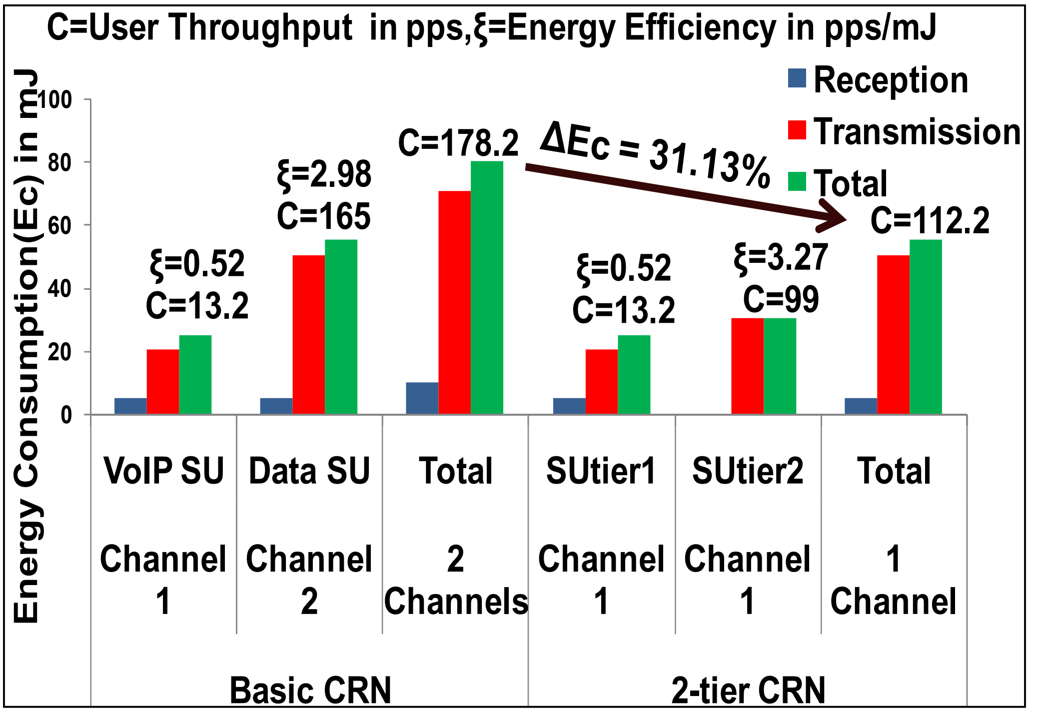


Fig. 6.33 Energy Consumption and Throughput for both VoIP and DATA SUs in Basic and 2-tier CRN

It is seen in Fig. 6.33 that for 0.3 ms sensing time and 3ms transmission time, energy consumed is far less for 2-tier CRN (55.25mJ) than Basic CRN (80.27mJ) with 31.13% decrease in energy consumption. Also, while energy efficiency [6.7] (in terms of throughput over energy consumption) of VoIP SUs is same in Basic and 2-tier CRN (0.52 pps/mJ), it is more for DATA SUs in 2-tier CRN (3.27 pps/mJ) than Basic CRN (2.98 pps/mJ). This gain in energy efficiency of DATA SUs occurs at the cost of loss in their throughput in 2-tier CRN (112.2 pps) than Basic CRN (178.2 pps), This is because while DATA SUs transmit for the entire duration in Basic CRN, 2-tier CRN allows them only the idle time slots of SUtier1 to transmit.

WARP Display unit records 11,527 and 68,538 transmitted packets for SUtier1 and SUtier2 respectively, depicting increase in system throughput for 2-

tier CRN. Moreover, Fig. 6.34 records a rise in spectrum efficiency [6.2] of 2-tier CRN (5.1 pps/Mhz) as compared to Basic CRN (4.04 pps/MHz), with 26.23 % increase in spectrum utilization. This results in increase in the total system throughput for 2-tier CRN as shown in Fig. 6.35.

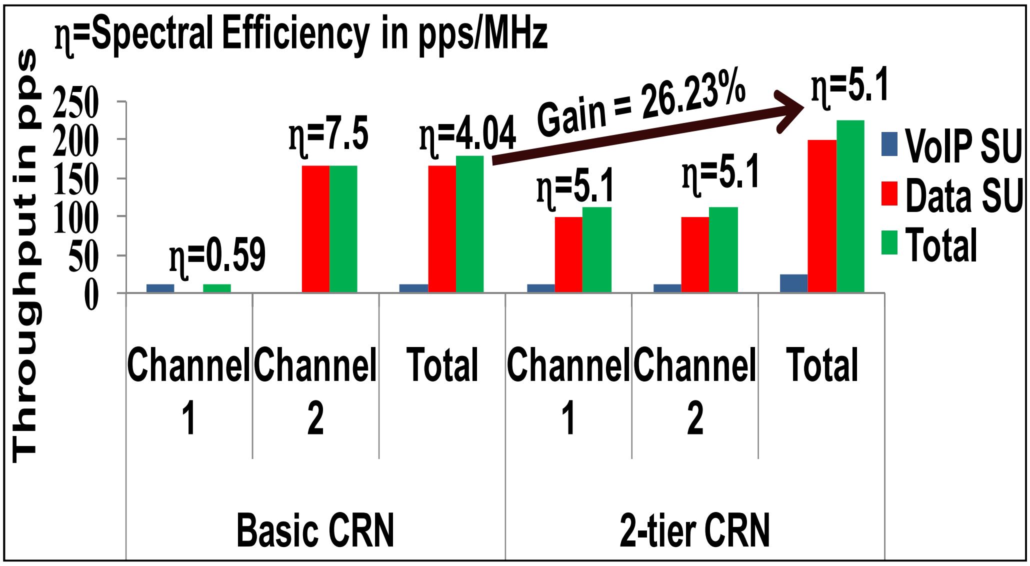


Fig. 6.34 Increase in Spectral Efficiency with respect to two channels in VoIP based 2-tier CRN over Basic CRN

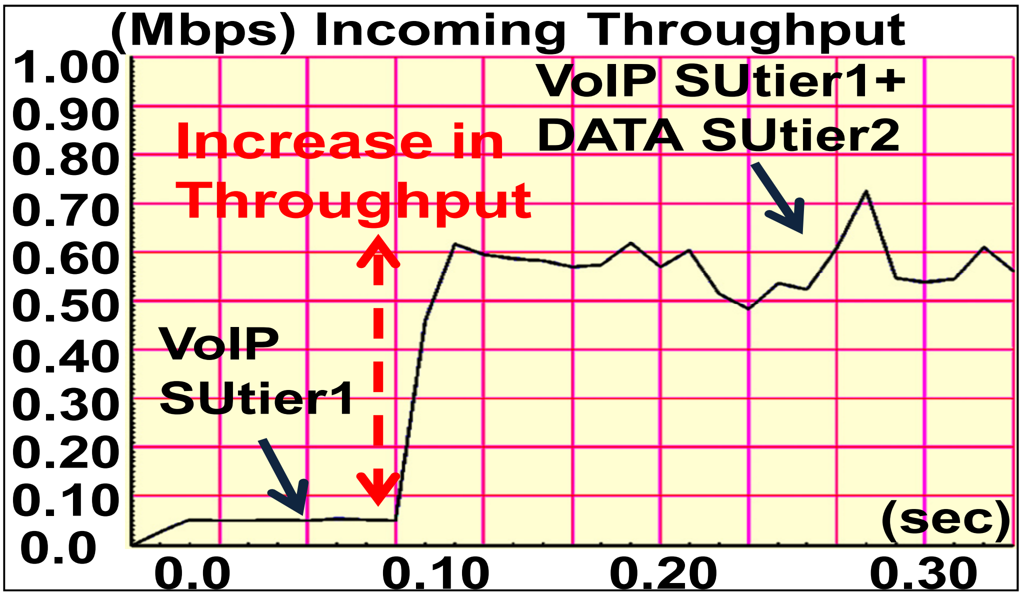


Fig. 6.35 Increase in System Throughput for 2-tier CRN

Thus, VoIP based 2-tier CRN fulfils the objectives in disaster management system, as it maintains good quality VoIP call and allows data transmission on the same channel.

It is, finally, inferred that VoIP based 2-tier CRN increases the overall spectrum utilization while maintaining the QoS of VoIP calls. In this regard, Table 6.5 summarizes the output generated from the designed models in this chapter.

Table 6.5 Summary Of Output Generated In Various Models

Designed Models	Significance	Important System Parameters	Basic CRN	VoIP based 2-tier CRN	Impro-vement	Comments
Mathematical Model	Design analytical framework for 2-tier CRN	SU Sum Goodput	39.6 pps	79.2 pps	100 %	Minimum increase in system capacity for single tx. by SUtier2 in 1 idle time slot
Markov Model	Study of different system probabilities	Successful Tx. Probability of 1 SU	0.45	0.54	20 %	As prob. of successful tx. increases for a single SU, system throughput increases.
Simulation Model	Validate the Mathematical Model and capture the complexity of 2-tier CRN in a multi-user multiple channel scenario	Through-put (Time Average)	3.09 pps	11.19 pps	262 %	Increase in system capacity for multiple tx. by SUtier2 in 1 idle time slot of SUtier1 under identical traffic rates of all SUs
		Link Utilization (Time Average)	10 %	40 %	300 %	
Test-bed Model	Practical Utility of the proposed system	Through-put	33.3 pps	198 pps	494 %	Actual increase in system capacity for different traffic rates of SUtier1 and SUtier2

6.8 Summary

This work has addressed the problem of limited spectrum utilization by SUs in Basic CRN with the design of a novel VoIP based 2-tier CRN that allows a second tier of DATA SUs to operate along with the VoIP SUs in the first tier. Rigorous stochastic analysis in mathematical framework confirms a minimum increase of SU Sum Goodput by 100% in 2-tier CRN under identical SU traffic rates. Thereafter, the complexities of 2-tier CRN in a multi-user multiple channel scenario are studied extensively in OPNET based real life-like simulation models, which illustrate an increase of 300% in link utilization for 2-tier CRN and also validates the analytical output.

In addition, Markov Models have been designed to highlight the difference between Basic and 2-tier CRN. Spectrum handoff has also been incorporated in the developed Markov Model for performance enhancement. Analysis of the Markov models has recorded significant reduction in SU dropping and blocking probabilities in spectrum handoff enabled 2-tier CRN with over 200% increase in successful transmission probabilities for SUs. A mathematical framework to study the SU behavior has been formulated, that has recorded the highest SU throughput after enabling spectrum handoff in 2-tier CRN and has, thus, confirmed the inference drawn from the Markov models.

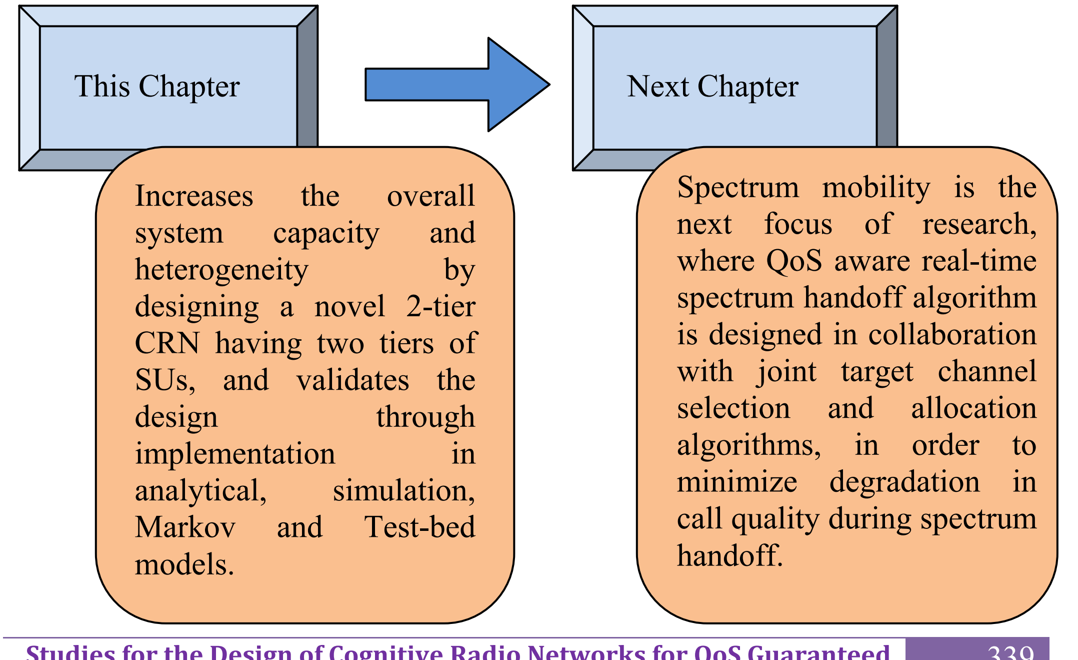
Moreover, the practical issues involved in deploying 2-tier CRN are discussed in this chapter. In this regard, two implementation algorithms namely, *Simple_msg* and *Periodic_msg* are designed to allow transmission by SUtier2 in the idle periods of SUtier1. Mathematical formulation of these algorithms is performed along with performance evaluation in analytical and simulation models. It is concluded that *Simple_msg* strategy provides higher transmission time for SUtier2 at the cost of increased time of interference and is suitable in scenarios with low SUtier1 activity. *Periodic_msg* algorithm incorporates periodic checking of channel status to reduce interference and is implemented when SUtier1 activity is high.

Finally, implementation in real test-bed is performed to demonstrate the practical utility of 2-tier CRN by deploying it in a disaster management system. The test-bed model successfully executes VoIP and data applications, and captures a rise of over 450% in system throughput for 2-tier CRN under different traffic rates, thus validating the proposition for practical applicability. Whereas, both system capacity and heterogeneity are increased in the proposed design of VoIP based 2-tier CRN, the system model is being studied further to devise compatible MAC protocols and spectrum handoff schemes apart from modifications in architectural issues.

The proposed 2-tier CRN system initiates new developments in research such as, *categorization of SUs* for the first and second tier of CRN, *design of MAC Protocols* to address issues of SU assignment in both tiers and manage the interaction among them, and subsequent *design of spectral handoff policies* to allow SUs perform handoff in both tiers on sudden arrival of PU. *Architectural issues* must be addressed with added functionalities in the spectrum broker node to maintain proper coordination in between the servers of the first and second tier of SUs. The most important part of the 2-tier based design of CRN is *determining optimal sensing and transmission intervals* for the SUs, along with *introduction of sensing algorithms* to detect idle periods of SUs. Thus, from the systems perspective, *2-tier CRN* is not only a mere enhancement of 1-tier Basic CRN, but altogether, it provides a new dimension in research.

The outcome of this work has been published in IEEE Systems Journal'16 (SCI Indexed) and IJACSA Journal'13 (SCI Indexed), and also published in the International Conference Proceedings of IEEE COMNETSAT'14 (Indonesia).

Also, the practical implementation aspects of the proposed concept with all its intricacies and methods are currently being drafted for patent application.



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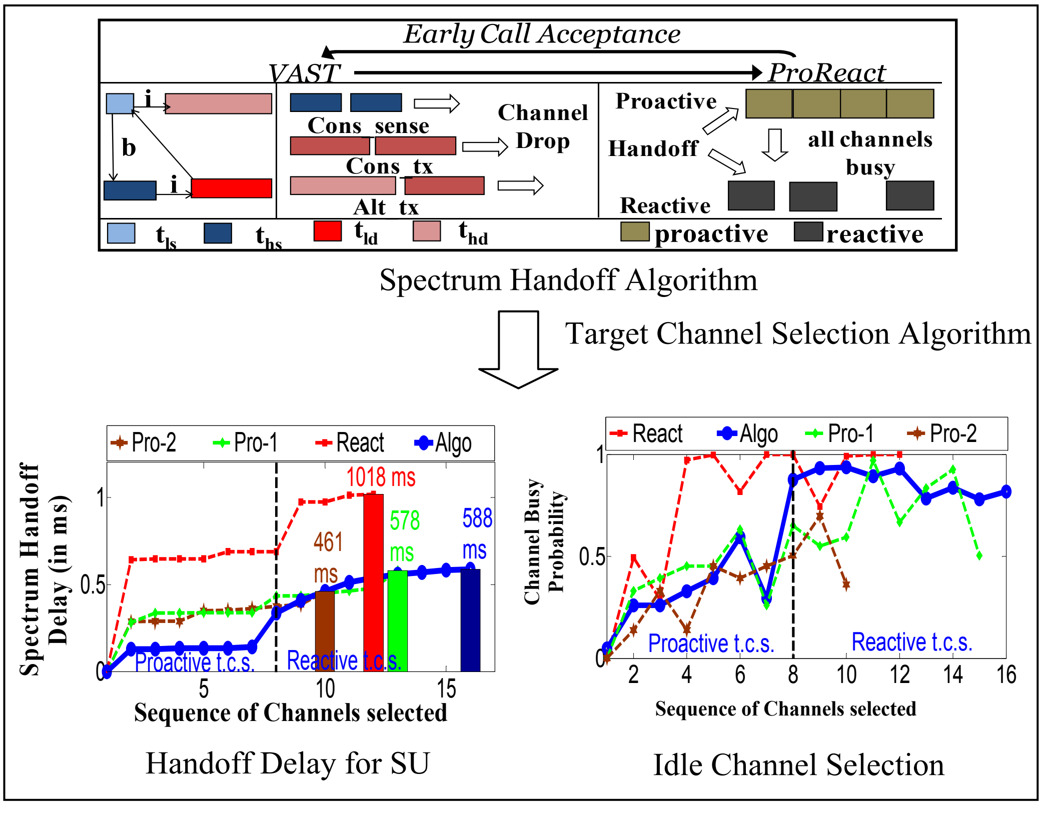
Chapter 7.

REAL-TIME SPECTRUM HANDOFF ALGORITHM WITH SUITABLE CHANNEL SELECTION STRATEGIES

Chapter Highlights



CHAPTER 7: Real-time Spectrum Handoff Algorithm with Suitable Channel Selection Strategies

"I'm a great believer that any tool that enhances communication has profound effects in terms of how people can learn from each other, and how they can achieve the kind of freedoms that they're interested in."

- Bill Gates, Microsoft

Outline of the Chapter

An ongoing VoIP communication by SU can be severely disrupted by the untimely presence of PU in the licensed channel. This implies that the SU must halt VoIP transmission, vacate the current channel and perform spectrum handoff at the earliest to a suitable idle channel to resume communication. In such a scenario, the spectrum handoff operation [7.1] must be jointly devised with the sensing, transmission and target channel selection mechanisms to form an integrated system whose overall aim is to sustain the VoIP communication for long periods amid varying network dynamics. This Chapter therefore shifts the focus of study from spectrum management policies in the previous chapters to the Spectrum Mobility aspect, which is an integral part of CRN. Although spectrum handoff has garnered much attention through numerous research activities, only limited ones have considered QoS sensitive applications during handoff design. This is a highly challenging and relevant problem in the CRN domain. Obviously, the studies of VoIP traffic over CRN will not be complete unless the spectrum handoff issues are addressed thoroughly.

In this regard, the contribution of this chapter includes i) the design of QoS aware spectrum handoff algorithm for each VoIP SU encompassing critical CR activities such as adaptive channel detection and transmission, three-level dropping decision, two-phase spectrum handoff operation and early call acceptance functions; and ii) the design of the Spectrum Controller (denoted by SC) node that performs channel estimation, target channel selection and channel allocation functions under a threshold time limit.

The operations of SC and SUs are closely tuned to each other with minimum latency and algorithmic complexity to preserve both power and time. Extensive performance analysis in analytical and simulation models confirms a decrease in spectrum handoff delay for VoIP SUs by more than 40% and 60%, compared to existing proactive and reactive algorithms, respectively and ensures a minimum 10% reduction in call-dropping probability with respect to the previous works in this domain. The effective SU transmission duration is also maximized in this process, rendering it suitable for successful VoIP communication.

7.1 Introduction

Spectrum mobility [7.2] is a significant aspect of CRN, where two crucial aspects gain prominence, namely, i) intelligent decision mechanism to drop from the current channel without interrupting PU transmissions, and ii) efficient channel selection algorithm to perform handoff without incurring significant delay, as illustrated in Fig. 7.1. All these operations must be executed in tandem to avoid QoS degradation and subsequent call drop. This requires an efficient design of real-time spectrum handoff methodology with the objective of minimizing the handoff delay and maximizing the probability of securing an idle channel.

Although considerable research has been done with respect to both proactive handoff (where handoff occurs to a predetermined target channel) and reactive handoff (where on-demand channel selection takes place) techniques, very few works [7.3, 7.4] have focused on designing spectrum handoff functionalities for real-time SUs in CRN. Accordingly, an in-depth review of the existing handoff related studies in the literature is carried out in this section.

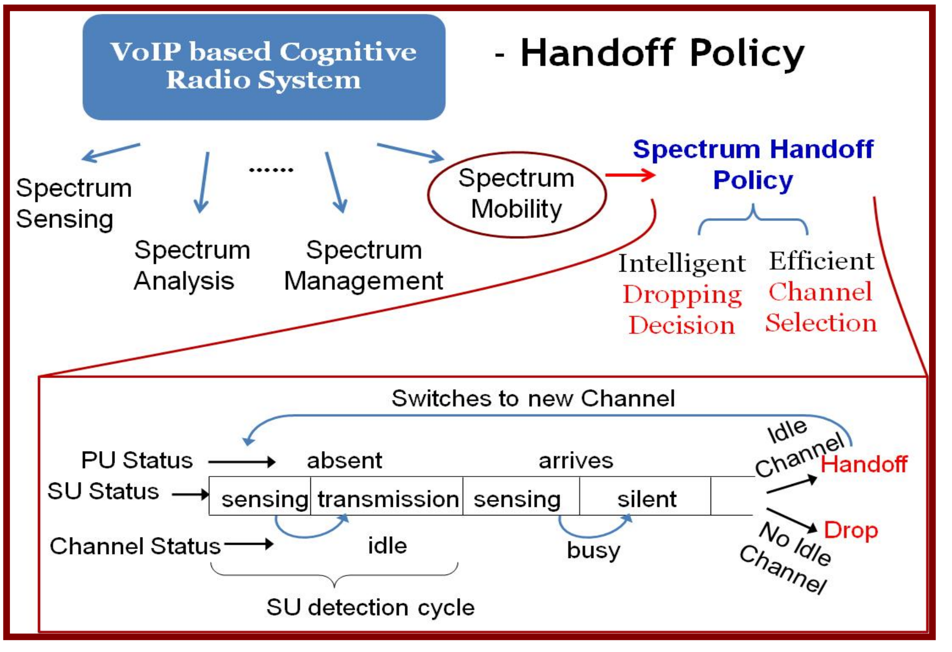


Fig. 7.1 Illustration of Spectrum Mobility in Cognitive Radio Networks

7.1.1 Literature Survey

Spectrum mobility primarily encompasses two important aspects, namely i) target channel selection, and ii) spectrum handoff design. Accordingly, this section provides an outlook of the ongoing research in these domains and subsequently, highlights the novelty of the work in this Chapter.

(i) Literature Survey: Target Channel Selection

Target Channel Sequence (TCS) calculation is performed on a proactive basis in [7.5], selecting channels with longer vacant time duration under Poisson arriving PUs. [7.6] selects the best TCS from a collection of six sets with minimum cumulative handoff delay using Dynamic Programming and Greedy approaches and also includes the “stay” policy in its handoff methodology. This work is extended in [7.7] which requires only five TCS sets to choose from. Channel idle probability and average waiting time in PRP M/G/1 queuing model are used to calculate the TCS in [7.8] that finally selects channels whose connection re-disruption probability is less. Finally, [7.9] considers two approaches namely equiprobable TCS and PU traffic intensity based TCS calculation and uses target channel selection probability for selecting the next idle channel. *TCS design in these works has not taken into consideration real-time application requirements such as for VoIP*. On the contrary, this chapter *selects QoS aware TCS with a fixed upper bound on the maximum estimated handoff delay*, while ensuring timely availability of a target channel during handoff by the SU.

(ii) ***Literature Survey: Spectrum Handoff Algorithm***

Apart from suitable target channel selection, spectrum handoff also includes a critical aspect - the decision to drop from the current channel which again can either be proactive (drop even before PU arrives) or reactive (drop after detecting PU presence). Accordingly, literature survey is carried out with respect to four possible scenarios as highlighted in the Table. Majority of these works has focused on either Proactive drop-Reactive Target Channel or vice-versa owing to their merits over other scenarios. However, there is a trade-off between handoff probability (and associated delay) and idle channel selection probability between these works. Additionally, limited research as in [7.3, 7.4] has studied VoIP applications during spectrum handoff design. It is observed that spectrum detection and transmission are not yet considered during algorithm design in the aforementioned works. This chapter, on the contrary, takes into account *both the real-time QoS requirements for VoIP calls as well as the sensing and transmission intervals* and *designs an integrated spectrum handoff algorithm* that combines the advantages of both proactive and reactive approaches.

The novelty of the work in this chapter is ascertained by the fact that the proposed design methodology involves the design of an integrated spectrum handoff algorithm combining proactive and reactive approaches along with QoS aware TCS mechanism specifically designed for VoIP applications, and the entire design is validated using analytical and simulation studies.

Table 7.1 Different Handoff Strategies: Pros and Cons

	Proactive Target Channel	Reactive Target Channel
Existing works on Proactive Drop Strategy [7.10-7.14]	• Channel search time • Interference with PU 🟩↓ • Channel utilization and idle channel selection probability in dynamic CRN 🟥↓ • Handoff probability and delay 🟥↑	• Interference with PU in dynamic CRN 🟩↓ • Idle channel selection probability in dynamic CRN 🟩↑ • Channel search time • Handoff probability 🟥↑ • Channel Utilization 🟥↓
	Research Works: [7.10]	Research Works: [7.11-7.14]
Existing works on Reactive Drop Strategy [7.5, 7.6],[7.15-7.17]	• Channel search time • Handoff probability 🟩↓ • Channel Utilization 🟩↑ • Interference with PU in dynamic CRN 🟥↑ • Idle channel selection probability in dynamic CRN 🟥↓	• Idle channel selection probability in dynamic CRN 🟩↑ • Channel utilization in dynamic CRN 🟩↑ • Handoff probability and delay 🟩↓ • Channel search time • Interference with PU 🟥↑
	Research Works: [7.5,7.6], [7.16,7.17]	Research Works: [7.15]
Proposed Work in this paper	• Dropping and handoff probabilities • Handoff delay • Interference with PU • Channel search time 🟩↓ • Idle channel selection probability • Channel utilization • Call quality (Application perspective 🟩↑	
• Positive Increase 🟩↑ • Positive Decrease 🟩↓ •Negative Increase 🟥↑ • Negative Decrease 🟥↓		

7.1.2 Significant Contributions

Driven by the lack of QoS support during spectrum handoff for VoIP based CRN as found in the literature survey, this chapter devises an integrated real-time spectrum handoff algorithm along with joint channel selection strategies, specifically related to VoIP applications for SUs and establishes the feasibility and superiority of the proposed design through analytical and simulation studies. The significant contributions of this work can be summarized as follows.

1. Novelty and motivation of the joint target channel selection and spectrum handoff algorithms for VoIP applications over CRN are established following an in-depth literature survey in Section 7.1.
2. The system model is developed in Section 7.2 along with necessary design considerations that are supported by suitable justifications.

3. In Section 7.3, the integrated real-time spectrum handoff algorithm is designed in three segments for each individual SU and their advantages are subsequently highlighted.
4. Next, the design methodology is provided for each SC node in Section 7.4, including the formulation of target channel selection problem and its calculation, subsequent channel allocation operations and coordination with the SUs.
5. The detailed mathematical framework behind the proposed design is established in Section 7.5 along with the derivation of critical system metrics (handoff and dropping probabilities, handoff delay, SU throughput and packet loss, etc.)
6. Comparative performance evaluation and superiority of the proposed algorithms is studied in simulation models in Section 7.6 that not only validate the proposed design but also confirm drastic performance improvement over existing works.

Finally, the chapter is concluded in Section 7.7.

7.2 Overview of the CR System Model

An overview of the system model for CRN to be considered is provided in this section, along with the necessary design considerations supported by proper justifications.

7.2.1 Design Considerations

1. A time-slotted CRN model [7.18] is considered, where every VoIP SU performs PU detection at the beginning of each time slot. Only when the sensing outcome is “idle”, SU transmits VoIP packets over the selected channel.
2. The SC Node comprises of two transceivers, whereas every VoIP SU is equipped with a single transceiver. This is justified since the power requirements at the Controller node are not as stringent as in the case of individual SUs.

3. The PUs are completely oblivious of SU presence in the respective channels. Therefore, the PU Controller may, in all probability, assign a channel under active SU transmissions to the incoming PUs. Thus, the CRN as considered in this work is a generic one.

4. Every PU is allotted its own dedicated channel. Once the channel is marked as a "candidate channel" for SU by the Spectrum Controller, only one VoIP communication between two SUs is allowed in that channel for the entire duration of pre-determined transmission duration.

7.2.2 Model Overview

The CRN comprises of one or more SC nodes that manage multiple SUs within their range as denoted in Fig. 7.2. The VoIP SUs under the supervision of an SC are marked as Real-time (RT) users so as to prioritize their transmissions over Non Real-time (NRT) users. Unless explicitly mentioned, any SU henceforth mentioned in this chapter refers to RT VoIP callers. Each of these SUs is equipped with a single transceiver and performs adaptive sensing and transmission operations using novel algorithms as explained in the later sections. Based on the dropping decisions, the SU performs real-time spectrum handoff to a predetermined target channel without terminating the call.

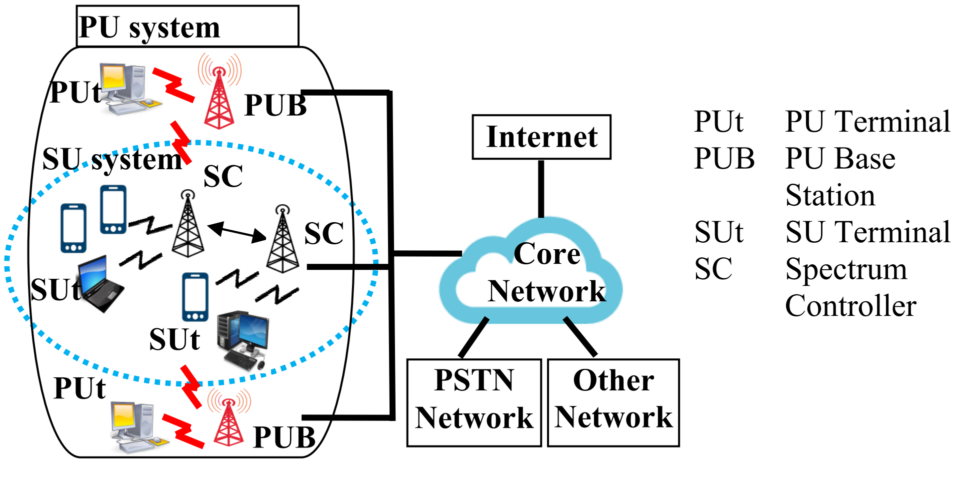


Fig. 7.2 Overview of the Proposed System Model comprising of PUs, VoIP SUs, PU Base Stations and SC nodes

Every SC is equipped with two transceivers and performs channel prediction, target channel calculation and channel allocation for the subscribed SUs under its domain. While one transceiver is used to sense and determine channel busy/idle characteristics, the other transceiver is utilized to communicate with the SUs via a dynamically updated Common Control Channel (CCC) [7.19]. It must be noted that while both SUs and SC operate concurrently, they are tightly coupled with each other as per the design methodology and must jointly satisfy two critical constraints namely; i) minimize the overall delay in VoIP communication during call initiation, normal communication and handoff operations; and ii) mitigate any interference and subsequent packet loss with the licensed PUs.

7.3 Real-time Spectrum Handoff Algorithm

Every SU involved in VoIP communication must adapt itself to varying channel conditions as prevalent in a CR network while still ensuring adequate QoS for the VoIP calls. Accordingly, a SU must perform three significant operations, that include i) allotting maximum transmission time to VoIP packets without interfering with ongoing PU activity, ii) timely decision to drop from the existing channel when either the call quality or the channel conditions deteriorate, and iii) successful spectrum handoff with minimum delay to a new channel without dropping the existing call. Equipped with a single transceiver, the SU must perform these operations with low algorithmic complexity and energy consumption. This section focuses on this innovative design aspect and develops an integrated real-time spectrum handoff algorithm that involves adaptive tuning of sensing and transmission intervals, three-level dropping decision policies, two-phase spectrum handoff strategies and early call acceptance policies.

The proposed algorithm is comprised of three segments. The first segment, *VAST* (VoIP based Adaptive Sensing and Transmission) deals with sensing, transmission and dropping decisions. This is followed by execution of the second segment, *ProReact* (Proactive and Reactive Handoff) which performs successful channel handoff to a new idle available channel without degrading the QoS limits (delay, loss) of VoIP calls. Finally, the *Early Call Acceptance* Policy is proposed in the third segment for quick resumption of VoIP calls in the selected channel. All these segments are discussed as follows.

7.3.1 *1st Part: VAST* (VoIP based Adaptive Sensing and Transmission)

This segment initially performs dynamic configuration of the sensing and transmission intervals as detailed in the following steps.

Proposed VAST Policy	
Step 1:	Two time intervals are considered each for channel sensing and transmission. The high duration sensing and transmission periods are denoted by t_{hs} and t_{hd} respectively. Similarly, the low duration sensing and transmission periods are denoted by t_{ls} and t_{ld} respectively.
Step 2:	Every CR cycle is defined as being comprised of one or more sensing intervals, followed by a single interval of successful SU transmission. Intuitively, the end of a transmission interval (t_{ld} or t_{hd}) marks the onset of a new CR cycle.
Step 3:	The SU performs energy based detection. If the detected energy is greater than certain predetermined threshold energy, the channel is considered busy, otherwise it is idle.
Step 4.	At the onset of every CR cycle, the SU senses the channel for t_{ls} duration. If the channel is sensed idle, it enters the t_{hd} period and transmits during this interval.
Step 5.	If the channel is sensed busy, the SU re-senses the channel for t_{hs} period. This is done to ensure that the channel busy alert is genuinely due to PU arrival, and not because of imperfect channel sensing. If the channel is again detected busy during this t_{hs} period, SU is aware of PU presence in the current channel and abstains from any further transmission.
Step 6.	If the SU senses the channel as idle during the t_{hs} duration, it enters the t_{ld} transmission interval to resume communication.

This policy is illustrated in the flowchart below.

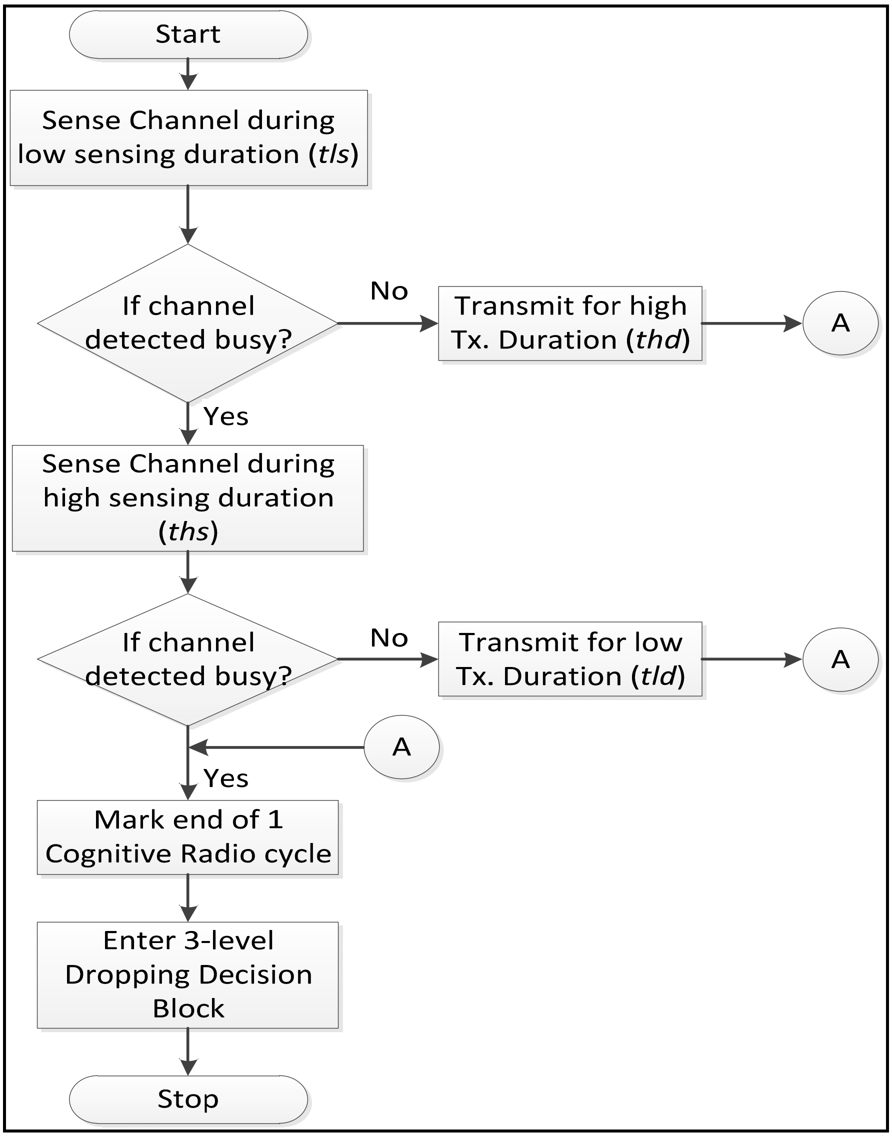


Fig. 7.3 Flowchart describing the *VAST* segment

Two important aspects are considered during execution of this policy, keeping in mind the power constraints and QoS requirements.

1. *Fast Recycling*: Once the channel is detected busy during an ongoing sensing cycle, the SU quickly switches to the next CR cycle instead of waiting for the rest of the sensing period. This saves both the energy consumption and subsequent sensing time.

2. *Continuous Sensing*: There is no time gap between t_{ls} and t_{hs} intervals. Although this consumes more power, it ensures that channel detection and subsequent dropping decisions are performed at the earliest to reduce associated delays during VoIP transmission.

In the second step, based on these four timing parameters (t_{ls}, t_{hs}, t_{ld}, t_{hd}), the *VAST* segment takes intelligent decisions to drop from the current channel. This step is very crucial because any drop from the current channel will lead to unwanted disruptions in communication. At the same time, retaining transmission in the presence of PU will lead to interference. The *Three-level dropping decision* policy as proposed in this work comprises of the following steps.

Three-level Dropping Decision Policy				
Level 1.	*Cons_sense*	(drop due to continuous sensing)	:	Drop from the current channel if this channel is sensed busy for two consecutive t_{hs} sensing periods in a single CR cycle.
Level 2.	*Cons_tx*	(drop due to consecutive low transmission periods)	:	Drop from the current channel if SU enters two t_{ld} transmission slots in consecutive CR cycles.
Level 3.	*Alt_tx*	(drop due to alternative high and low transmission periods)	:	Consider n_t_{ld} and n_t_{hd} as the total number of t_{ld} and t_{hd} intervals respectively, entered for transmission by the SU, within a span of n_{cycle} CR cycles. Accordingly, drop from current channel if n_t_{ld} is $\geq (1/2)\, n\_t_{hd}$.

All the three dropping decisions effectively cater to diverse network behavior. *Cons_sense* is the reactive dropping decision that ensures that SU does not stay on the current channel (already occupied by a PU) to resume communication, thus reducing the handoff delay. *Cons_tx* is the proactive decision technique that prevents SUs from transmitting in shorter time intervals, as this leads to frequent packet loss and deteriorates the call quality. *Alt_tx* is another proactive approach that detects irregular time durations and prevents any VoIP call under such situation (as this introduces jitter [7.20] and degrades VoIP QoS). The *Three-level dropping decision* policy is represented in Fig. 7.4.

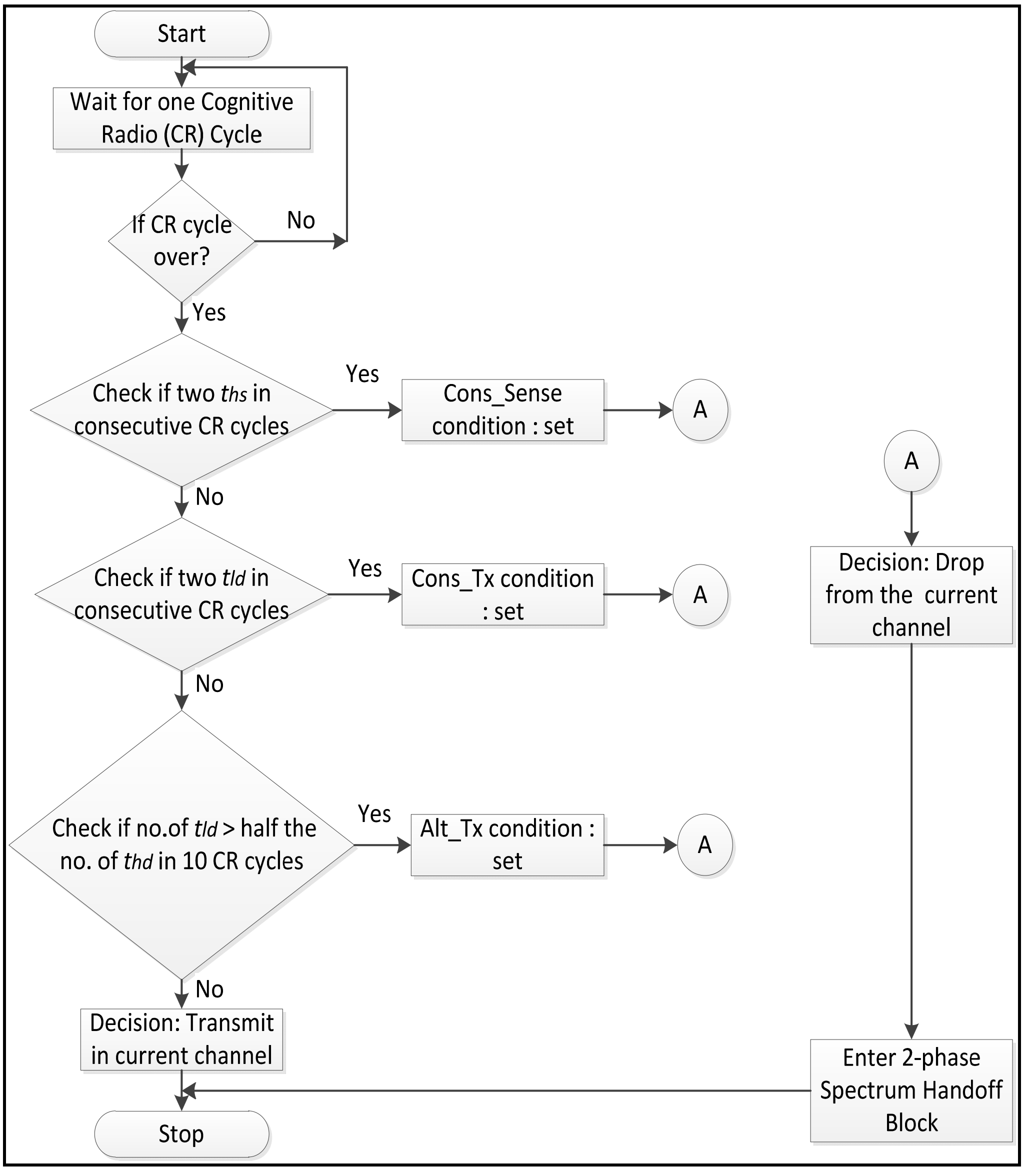


Fig. 7.4 Flowchart explaining the *Three-level dropping decision* policy

7.3.2 *2nd Part: ProReact* (Proactive and Reactive Handoff)

As SU drops from the current channel on executing *VAST* segment, it must find an idle channel at the earliest and resume communication. This is performed by the two-phase spectrum handoff strategy *ProReact* in the second module of our proposed algorithm. *ProReact* comprises of two phases as explained below.

(i) ***Proactive Phase: Perform handoff proactively to a pre-determined target channel***

In the *Proactive* phase, every SU periodically obtains a pre-determined TCS from the SC node. After dropping from the current channel, the SU scans every channel in the TCS sequentially and on finding any available idle channel resumes transmission. The SU sends *l_pro* message via the CCC to the SC and informs it about the currently allocated channel. In response, the SU obtains the updated TCS from the SC to be used in the next handoff process. The TCS should be so designed by the SC such that the total time incurred in sensing all the channels in the TCS must not exceed a certain delay limit.

(ii) ***Reactive Phase: Perform handoff reactively to a reactive target channel on an on-demand basis***

In the worst case scenario, when all channels in the TCS are sensed busy, the SU enters the *Reactive* Phase of *ProReact* segment, where it requests the SC for reactive target channels by sending *l_react* message using the CCC. On receiving the list of target channels, the SU performs reactive handoff by randomly and uniformly selecting any available channel for sensing and subsequent transmission if that channel is idle. This is done to ensure that the call is not dropped. However, the total number of channels (denoted by $n_{maxsearch}$) to be sensed reactively is limited by the maximum handoff time as allowed by SU.

It is worth mentioning that the *Proactive* phase corresponds to the best and average case scenarios, where the network behavior does not vary abruptly and the TCS is updated periodically. *Reactive* phase belongs to the worst case, where variation in network parameters occurs even before TCS is updated by the SC. Obviously, the probability of SU entering the *Reactive* phase is much lower compared to its *Proactive* counterpart.

The flowchart in Fig. 7.5 describes the operations of the *ProReact* segment.

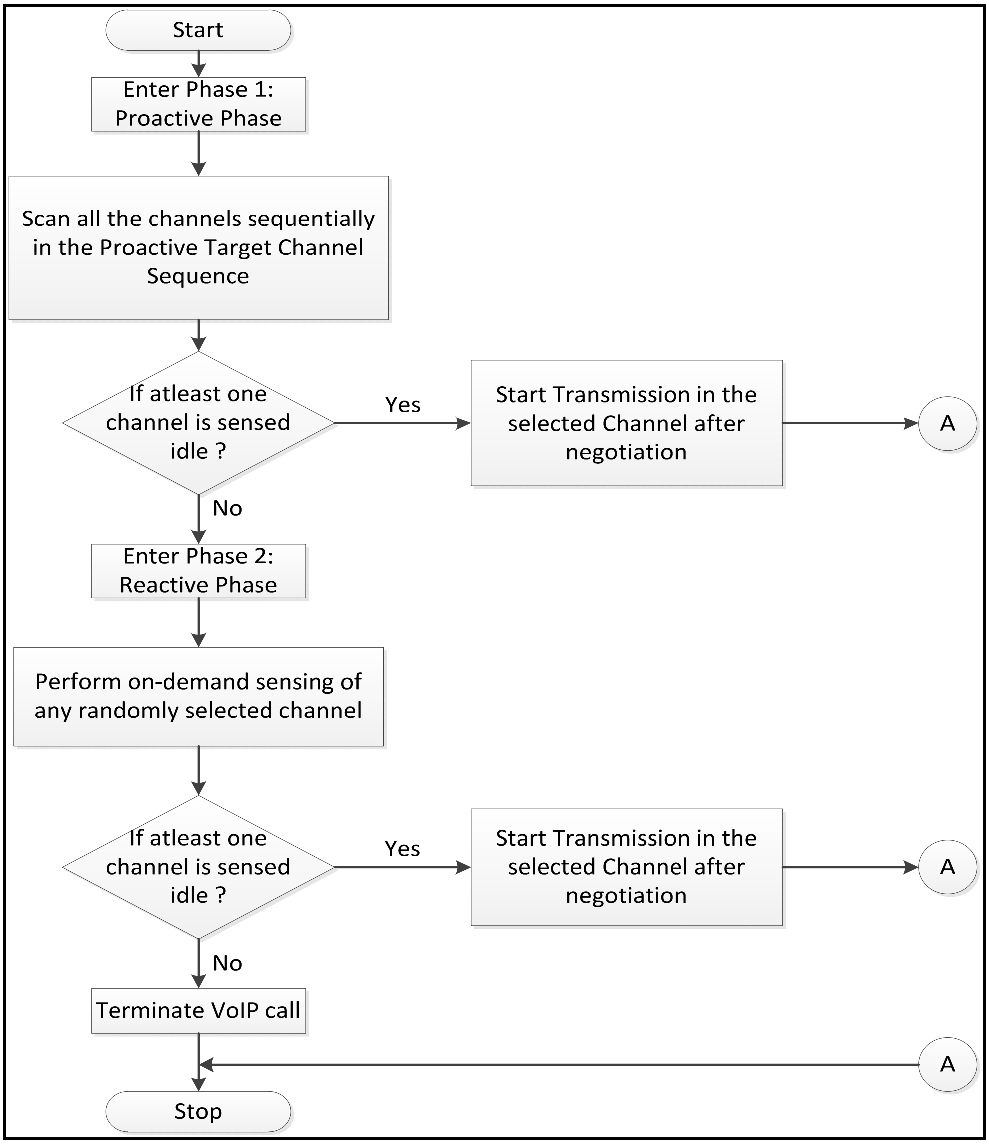


Fig. 7.5 *ProReact* Segment demonstrated using Flowchart

7.3.3 *3rd Part: Early Call Acceptance* Policy

Following successful spectrum handoff, the interrupted VoIP call must be resumed at the earliest to prevent any call drop. This is because a VoIP call can be terminated if call timeout occurs during the call setup time (~3 s for H.245 as per H.323 protocol [7.21]) and also during RTCP message exchanges. Hence, the *Early Call Acceptance* Policy forms the third segment in our designed algorithm and is proposed below.

Early Call Acceptance Ratio

1. Enable VoIP early start mechanisms as part of the underlying call signaling protocol. Considering H.323, this implies the activation of VoIP fast start, early H.245 and H.245 Tunneling modes.

2. Reduce the initial sensing time t_{s_i} to a minimum value where $t_{s\_i} < t_{ls}$. This is done to avoid any time loss due to sensing, as the probability of the channel being idle is high both during channel allocation and *Proactive* handoff phases.

3. When the SU senses the first channel in TCS as busy during the t_{s_i} period, it waits for a short interval of 50 ms before performing the usual t_{ls} - t_{hs} operations and repeats this step twice before moving on to sense the next channel. This is done because based on the TCS calculation by the SC, the PU may be active on that channel for a short time interval although with a very low probability. Continuous sensing is avoided keeping in mind the power constraints of individual SUs. The maximum sensing time in that case is equal to t_{s_i} +2*(50+t_{ls}+t_{hs}).

7.3.4 Advantages of the Proposed Algorithm

All the three segments in the proposed algorithm operate concurrently to ensure the following benefits for VoIP SUs in CRN.

1. Adaptive sensing and transmission durations minimize the effects of imperfect spectrum sensing by SU.

2. *Cons_sense* reduces the probability of unnecessarily dropping from the current channel (and subsequent handoff delay) by using the reactive strategy to drop from the channel.

3. *Alt_tx* and *Cons_tx* reduce interference time (and subsequent packet loss) with sudden PU arrival by proactively vacating the channel when high busy times are recorded.

4. The *Proactive* handoff policy reduces the handoff delay, whereas the *Reactive* handoff strategy along with the *early call acceptance* policy ensures reduction in the call dropping probability.

The schematic diagram in Fig. 7.6 depicts the relationship among all the three segments of the proposed algorithm.

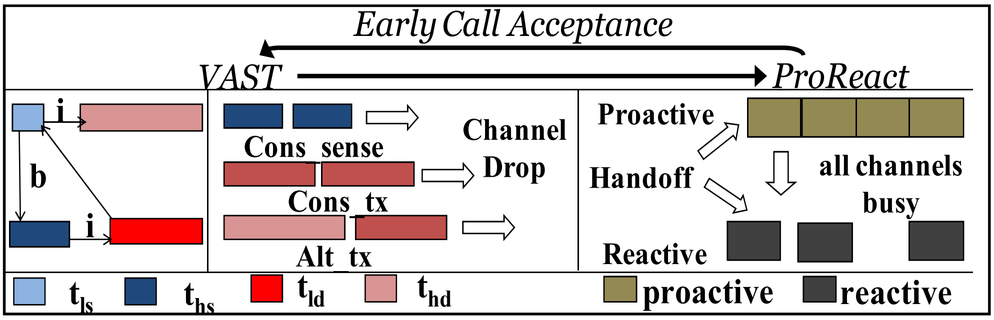


Fig. 7.6 Schematic Diagram illustrating the proposed Spectrum Handoff Algorithm in a nutshell

7.3.5 Requirement of the Spectrum Controller

The target channel selection and subsequent channel allocation in this work are performed not by individual SUs but by the SC because of two significant reasons.

1. With availability of high power and increased processing speed, the SC can not only sense a wide range of channels before selecting the TCS but also it can update the TCS in real-time. The same is not possible for a VoIP SU to perform with a single transceiver as it needs to dedicate maximum transmission time to its transceiver rather than sensing.

2. Also, if individual SUs are still allowed to calculate their own TCS (using different transceivers and high-speed processors which is not a practical and cost-effective solution), there is high probability that two different SUs may arrive at the same target channel, leading to collision and call drop. There are three possibilities namely; a) the SU can wait in a high-priority queue for an ongoing SU to finish transmission. However if both SUs are RT users, the waiting SU cannot preempt the transmitting SU and the waiting delay increases, which is not suitable for VoIP calls. b) The new SU can choose to shift to another channel for sensing. As number of SUs increase, such instances will increase. Thus the *Proactive* phase will

cease to exist and all SUs will perform *reactive* handoff which is not desirable. c) On the other hand, if the new SU performs target channel selection at every instance of handoff, the time complexity increases manifold which will ultimately render the VoIP call useless.

Accordingly, the SC not only selects and updates the TCS for all SUs but also manages their allocation using different vectors to mitigate this collision as explained in the next section.

7.4 Target Channel Selection and Allocation

It is evident from the previous section that every SU interacts with the SC during three phases; i) Proactive TCS Request/Response, ii) Periodic TCS Request/Response, and iii) Reactive TCS Request/Response. Accordingly, the SC node must perform three operations;

1. Estimate the channel idle/busy characteristics using the first transceiver and select the candidate channels;
2. Perform proactive TCS calculation on the candidate channels with minimum time and algorithmic complexity;
3. Communicate with the SUs and allocate selected channels (either Proactive TCS or Reactive TCS) to them using the second transceiver.

7.4.1 Channel Sensing and Selection

The SC performs the sensing of different frequency bands (denoted as channels) and estimates their average busy and idle periods after observing the channel usage characteristics. The mode of operation is as follows:

1. The first wireless transceiver sequentially senses each channel for a certain amount of time, based on which the SC calculates the average transition rates from idle to busy periods and vice-versa. Let $(\alpha,\beta)_{(\omega)}^{(t)}$ denote the average transition rates from idle to busy periods and vice-versa respectively for channel ω at time slot t.
2. Based on the observed values of $(\alpha,\beta)_{(\omega)}^{(t)}$, the SC calculates the estimated channel idle time and handoff time for every channel. Considering the

average VoIP talkspurt value of 270-300 ms, let the suitable channel idle time (denoted by t_{idle_ideal}) be atleast 300 ms for VoIP calls to sustain. Also, consider the suitable channel busy time (denoted by t_{busy_ideal}) to be 250 ms. This is justified because any channel whose busy time is over 250 ms is not suitable for QoS sensitive VoIP transmissions.

Let $fIR_{(\omega)}^{(t)}$ and $fBR_{(\omega)}^{(t)}$ be the probability density functions of the channel (ω) idle and busy time in the current time slot t respectively. The estimated channel busy and idle periods are given by,

$$t_{idle}^{(t)}[\omega] = \int_{0}^{t_{idle\_ideal}} fIR_{(\omega)}^{(t)} t_1 \, dt_1 + \int_{t_{idle\_ideal}}^{\infty} fIR_{(\omega)}^{(t)} t_{idle\_ideal} \, dt_1 \tag{7.1}$$

$$t_{busy}^{(t)}[\omega] = \int_{0}^{t_{busy\_ideal}} fBR_{(\omega)}^{(t)} t_2 \, dt_2 + \int_{t_{busy\_ideal}}^{\infty} fBR_{(\omega)}^{(t)} t_{busy\_ideal} \, dt_2 \tag{7.2}$$

where t_1, t_2 = time instants at which channel (ω) changes state from idle to busy and vice-versa respectively. In addition, the estimated channel handoff time comprises of the total channel busy period along with the channel switching time (denoted as t_{ch_sw}) and is expressed as,

$$t_{handoff}^{(t)}[\omega] = t_{busy}^{(t)}[\omega] + t_{ch\_sw} \tag{7.3}$$

3. Based on the observed values of $(\alpha, \beta)_{(\omega)}^{(t)}$ and considering a standard traffic model for the channel usage by the PUs, the estimated values of $t_{idle}^{(t)}[\omega]$ and $t_{handoff}^{(t)}[\omega]$ can be calculated to be used later in the TCS calculation. For example, considering an exponential on-off traffic model for PUs [7.22], the estimated idle channel and handoff periods are derived from (7.1) and (7.2) respectively and are given by (7.4) and (7.5) respectively.

$$t_{idle}^{(t)}[\omega] = \frac{1}{\beta_{(\omega)}^{(t)}} \left(1 - e^{-\beta_{(\omega)}^{(t)} \times t_{idle\_ideal}} \right) \tag{7.4}$$

$$t_{handoff}^{(t)}[\omega] = \frac{1}{\alpha_{(\omega)}^{(t)}} \left(1 - e^{-\alpha_{(\omega)}^{(t)} \times t_{busy\_ideal}} \right) + t_{ch\_sw} \tag{7.5}$$

where t_{idle_ideal} >=300 ms, t_{busy_ideal} <=250 ms. It must be noted that in this study, the channel state changing twice or more within a single duration is not considered because the probability of such an occurrence is very low within short t_{idle_ideal} and t_{busy_ideal} periods. Even if such a condition arises, that channel is not suitable for VoIP transmissions and will eventually be discarded by the SC. Any channel whose idle time is less than the busy period is also deleted from subsequent target channel calculations.

7.4.2 Proactive TCS Calculation for *Proactive* Handoff phase in SU

The SC performs TCS calculation considering two significant factors; i) the handoff delay for the sequence of channels for every SU must not cross a total handoff delay limit (denoted by $T_{handoff}$), and ii) the calculation process must be done with low algorithmic and time complexity. Accordingly, this section designs the SC node to perform TCS Calculation in two phases. Initially, the TCS Problem is devised as a Fractional Knapsack problem. Then the formulated problem is solved using Greedy approach based *GA_TCS* Algorithm.

(i) <u>TCS Problem Formulation: Fractional Knapsack Problem</u>

The Target Channel Selection problem is mapped to the standard Fractional Knapsack Problem [7.23] with minor modifications. Formally, the Knapsack Problem is stated as

"*Given a sack of capacity C, how many items 'i' each of size S_i can be fit into it, so that the value of the sack is maximized, where each item 'i' has a value v_i ?*"

In order to formulate the TCS problem, we substitute C with $T_{handoff}$, i by ω, S_i by $t^{(t)}_{handoff}[\omega]$ and v_i by $t^{(t)}_{idle}[\omega]$. Accordingly, the Fractional Knapsack problem is mapped as follows.

"*Given a total time limit of $T_{handoff}$ at time t, how many channels 'ω_i' each having an estimated handoff time $t^{(t)}_{handoff}[\omega_i]$ can be fit into the TCS set so that the value is maximized, where each channel ω_i is valued by its remaining idle time $t^{(t)}_{idle}[\omega_i]$?*"

Mathematically, the problem is defined as follows:

TCS Calculation Problem: Fractional Knapsack

Input: Non-negative integral vectors: $t_{handoff}^{(t)}[\omega_1,\omega_2,...]$, $t_{idle}^{(t)}[\omega_1,\omega_2,...]$, $\omega=[\omega_1,\omega_2,...,\omega_n]$; *Integer:* $T_{handoff}$

Objective: $\max_{j\in\omega} f(j)=\sum_{j=\omega_1}^{\omega_n} t_{idle}^{(t)}[j]\times x(j)$ *subject to*

$$\sum_{j=\omega_1}^{\omega_n} t_{handoff}^{(t)}[j]\times x(j)\le T_{handoff} \tag{7.6}$$

where $x(j)=1 \quad \forall\, \omega_1\le j\le \omega_{k-1}$,

$x(j)=0 \quad \forall\, \omega_{k+1}\le j\le \omega_n$

$$x(j)=\frac{T_{handoff}-\sum_{i=\omega_1}^{\omega_k} t_{handoff}^{(t)}[i]}{t_{handoff}^{(t)}[\omega_k]}\ \forall\ j=k$$

where *i)* $\frac{t_{idle}^{(t)}[\omega_1]}{t_{handoff}^{(t)}[\omega_1]}\ge\frac{t_{idle}^{(t)}[\omega_2]}{t_{handoff}^{(t)}[\omega_2]}\ge\cdots\ge\frac{t_{idle}^{(t)}[\omega_n]}{t_{handoff}^{(t)}[\omega_n]}$

ii) $\omega_k=\min\left[\omega_j\in\{\omega_1,\omega_2,.....,\omega_n\}:\ \sum_{i=\omega_1}^{\omega_j} t_{handoff}^{(t)}[i]>T_{handoff}\right]$

(ii) ***Greedy Solution: GA_TCS Algorithm***

GA_TCS algorithm is designed to solve the aforementioned problem using greedy technique and select the best TCS for each SU. Let *U* be the set of all SUs under one SC. Accordingly, the algorithm is proposed below.

GA_TCS Algorithm

Input: $T_{handoff}$, $t_{handoff}^{(t)}[\omega_1,\omega_2,...]$, $t_{idle}^{(t)}[\omega_1,\omega_2,...]$, $\omega=[\omega_1,\omega_2,...,\omega_n]$;

Step 1:	Set k=$T_{handoff}$, *flag_end*=0, *j*=1, *l_select*$^{(t)}$= ω;	// Initialize

Step 2:	For each ω_i, calculate $val(\omega_i) = \dfrac{t^{(t)}_{idle}[\omega_i]}{t^{(t)}_{handoff}[\omega_i]}$;	// Sort ratio in descending order of value/size ratio
Step 3:	Sort ω such that $val(\omega_i) \geq val(\omega_{i+1})$;	
Step 4:	Run Iteration till *flag_end*==1 { If (ω $\neq$ null) {	
Step 7:	If (k>=$t^{(t)}_{handoff}[\omega_i]$) { Include ω_i in $l^{(t)}_{pro}[\sigma_1, \sigma_2, \ldots]$ where $\sigma_j = \omega_i$, $j=j+1$; delete ω_i from ω; $k=k-t^{(t)}_{handoff}[\omega_i]$; }	// ω_i = suitable target channel by SU, so include it in TCS
Step 9:	Else If (k==0) { *flag_end*=0; }	// The Knapsack is full.
Step 13:	Else { Calculate $f=k/\ t^{(t)}_{handoff}[\omega_i]$;	// Calculate fraction
Step 15:	If ($f > 0.5$) { Include ω_i in $l^{(t)}_{pro}[\sigma_1, \sigma_2, \ldots]$ such that $\sigma_j = \omega_i$,	// ω_i = suitable candidate when the fraction > 0.5.
Step 16:	$j=j+1$; Delete ω_i from ω; flag_end=0 ; } } }	
Step 19:	Else { $\omega = l\_select^{(t)}$; } }	// Re-start from ω_1 in sorted ω

Output: $\left(l^{(t)}_{pro}[\ ]\right)_{Ui} = l^{(t)}_{pro}[\sigma_1, \sigma_2, \ldots]$ *where* $\left(l^{(t)}_{pro}[\ ]\right)_{Ui}$ = *Proactive TCS for User* U_i *at time slot t.*

The SC calculates the *GA_TCS* Algorithm for every user under its domain corresponding to every call-id as subscribed in the SC. However, it modifies the proposed algorithm under two situations. Firstly, during calculation of TCS for any user, the SC considers all the sensed yet unallocated channels except the first channel in the TCS of every subscribed SU. This is because on every successful proactive spectrum handoff by the SU to the first channel of its TCS, the TCS is again updated by the SC following *l_pro* message exchange. Secondly, if the total number of unique channels in the calculated TCS of any SU is less than three, that TCS is discarded and the corresponding VoIP call is dropped. This is because any TCS having less than three unique channels denotes that the total expected busy time of the three channels is equal to $T_{handoff}$. This implies that each of these channels has a relatively high busy period and, thus, is unsuitable for uninterrupted VoIP transmissions.

The SC deploys its first wireless transceiver to sense a certain number of channels for a predetermined sensing time, based on which it calculates the TCS for every SU and broadcasts the same on their operating frequencies on a periodic basis. In addition, the SC deploys its second wireless transceiver to receive any incoming message from SU on the CCC. On receiving any *l_pro* message from the U_ith SU, the SC halts periodic TCS calculation and sends the most recently updated TCS $\left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ to the U_i. However, the highest priority is given to the *l_react* message from any incoming SU for which the SC performs Reactive TCS operation as described later.

(iii) ***Proof of Correctness and Optimality of GA_TCS***

The proposed *GA_TCS* algorithm is now evaluated for correctness and optimality. Let the Greedy Solution for *GA_TCS* be denoted by $G=\{g_1,g_2,\ldots,g_k\} \subset \omega$. Let the optimal solution to the Fractional Knapsack problem for TCS calculation (denoted by P) be given by $O = \{o_1,o_2,\ldots,o_k\} \subset \omega$.

Step 1: Prove that there exists one optimal solution O′ such that it selects the item g_1 which is the first greedy choice as per the proposed solution.

Proof: Let G selects g_1 as its first channel. The idea is to prove that O will also select g_1 as the first channel such that $o_1=g_1$. Suppose O does not select g_1. Then the corresponding weight of g_1 (given by $t_{handoff}^{(t)}[g_1]$) is removed from O and g_1 is included in O' which is the new solution. Since O' also has the same weight constraint (as given by $T_{handoff}$), and since g_1 has the maximum value-weight ratio in terms of $\frac{t_{idle}^{(t)}[g_1]}{t_{handoff}^{(t)}[g_1]}$, O' is as good as O in yielding the maximum value (or else, there will be a contradiction). Hence, $g_1 \in O'$ is an optimal solution for P.

Step 2: Prove the optimal substructure property: "an optimal solution to the problem contains within it optimal solutions of subproblems".

Proof: Proceeding from *Step 1*, after g_1 is selected, the weight constraint reduces to $T_{handoff}'' = T_{handoff} - t_{handoff}^{(t)}[g_1]$. The channel list becomes $\omega'' = \omega - \{g_1\}$. Let P'' be the updated fractional knapsack problem whose weight constraint is K'' and channel list is ω''. Also let $O''=O'-\{g_1\}$.The idea is to prove that O'' is an optimal solution to P'' and thus exhibits optimal substructure property. Suppose O'' is not an optimal solution to P''. Let Q be the optimal solution for P'' such that it yields higher value (idle times) than O''. Let $R=Q \cup \{g_1\}$ which means R is a feasible solution to P. Again, the value of O' (an optimal solution for P) is equal to the value of $O''+ \{g_1\}$, which is less than the value of R because value of O'' is less than Q. Thus, R becomes more valuable than the optimal solution O' which is a contradiction. Hence, O'' is an optimal solution for P''.

Thus, combining Steps 1 and 2, after each greedy choice is made as an optimal solution, the new problem is of the same form as the original one and thus it is proved inductively that the solution based on *GA_TCS* algorithm is an optimal solution.

(iv) ***Advantages of the Proposed TCS Algorithm***

The primary reason for mapping the TCS as a Fractional Knapsack problem is to calculate the number of available idle channels within a certain delay bound, yet maximizing the channel idle time for VoIP sessions. Fractional knapsack is preferred over Integer (0/1) Knapsack as the latter is a NP-hard problem. Although the (0/1) problem can be solved using Dynamic Programming, the time complexity is pseudo-polynomial *(nC)* [7.23] which increases as the encoding of C changes from unary to binary. Therefore, this work chooses Fractional Knapsack which can be solved optimally using greedy algorithm with reduced time complexity of $O\ (n \log n)$. This means that the SC can provide faster updates of TCS for every SU. Finally, the value/size ratio is preferred over only values (maximization) and sizes (minimization) in *GA_TCS* algorithm to ensure the optimal selection of channels in the TCS.

Moreover, the proposed algorithm also ensures that the joint probability of the same channel being accessed as the target channel by two different SUs at the same time as given by the following expression is negligible.

$$\text{Probability that two SUs access the same channel at the same time} = \text{Probability that both SUs drop at the same time slot} \times \text{Probability that two SUs request for TCS using } l\_pro \text{ at the same time} \times \text{Probability that two SUs sense their first target channel (with the best idle/busy ratio) as busy} \qquad (7.7)$$

7.4.3 Reactive TCS Calculation for Reactive Handoff Phase in SU

As any SU sends *l_react* message to the SC in the *Reactive* handoff scenario, the SC immediately selects all the unallocated channels, forms a pseudo-random sequence of those channels and sends that list to the requesting SU on the CCC. The SU on its part further calculates the maximum number of channels that can be sensed within the threshold delay limit and, thereafter, randomly and uniformly selects any available channel for transmission.

It must be noted that although Reactive TCS calculation by SC incurs more delay compared to on-demand reactive channel search by individual SUs, the above process is preferred due to two reasons. Firstly, this condition occurs only in the worst case with a low probability under diverse network scenarios when the proactive handoff fails. Under such circumstances, random channel search by individual SUs within the threshold delay limit will yield imperfect outcomes, eventually leading to call drop. So requesting SC for the possible target channels reduces the channel search time and also increases the probability of finding an idle channel successfully. Secondly, when two SUs belonging to separate call-ids perform reactive handoff at the same instant, in absence of any SC, it may so happen that both the SUs select the same channel and start transmissions, eventually leading to collisions. This scenario is avoided by using pseudo-random sequence in the Reactive TCS calculation by the SC. The probability of two SUs requesting *l_react* to SC at the same time, however, is very low as it involves joint probabilities as expressed below.

Probability that two SUs send *l_react* at the same time	=	Probability that both SUs drop at the same time slot	X	Probability that both SUs find the first, second, third, …, last channel in TCS busy at the same time slot

(7.8)

7.4.4 Channel, User and TCS Allocation Vectors

For every SU involved in VoIP call, the SC maintains a unique session id (equivalent to the call-id) so as to execute *GA_TCS* algorithm corresponding to every session id (rather than users) and broadcast the calculated $\left(l_{pro}^{(t)}[\;]\right)_{Ui}$ to every participating SU (U_i) within the same session. Accordingly, the SC maintains three vectors to simplify its operations with respect to user and channel allocation. All these vectors are defined as follows.

(i) <u>Channel Allocation Vector:</u>

V contains details of all the channels selected as candidate channels by the SC and is defined as, $C\_V = \{(w_i, v_i): w = \omega\}$ where

$v_i = 0$ if w_i is not allotted to any SU

$= 1$ if w_i is included as the first in the TCS of any SU

$= 2$ if w_i is selected as the CCC

$= k$ if w_i is currently allotted to SUs belonging to session id $= k$

(ii) <u>User Allocation Vector:</u>

U_V contains the details of all the subscribed SUs under the concerned SC and is defined as, $U\_V = \{(u_i, x_i): u = U\}$ where $x_i = k$ if u_i belongs to session id $= k$.

(iii) <u>TCS Allocation Vector:</u>

T_V maintains the $\left(l_{pro}^{(t)}[\;]\right)_{Ui}$ for every user $U_i \in U$ and is defined as, $T\_V = \{(t_{i,1};\ t_{i,2})\}$ where $t_{i,1} = k$ for session id=k, $t_{i,2} = \left(l_{pro}^{(t)}[\;]\right)_{Ui}$ for every U_i belonging to session id $= k$.

(iv) Illustrative Example of the Allocation Vectors

Consider 9 channels and 5 SUs within a single SC. Let U_i denote *i*th SU. Let ω_6 be the CCC. Consider the following scenario in Table 7.2.

Table 7.2 Illustrative Example for Allocation Vectors

VoIP Call	User ID	Session ID	Operating Channel	TCS
1	U_1 and U_3	12112	4	$\{\omega_7, \omega_5\}$
2	U_2 and U_4	34112	1	$\{\omega_9, \omega_2\}$
3	U_5 (Other SU is within another SC)	12222	3	$\{\omega_2, \omega_8\}$

The different allocation vectors are calculated by the SC as follows.

1. $C\_V = \{(\omega_1, 34112), (\omega_2, 1), (\omega_3, 12222), (\omega_4, 12112), (\omega_5, 1), (\omega_6, 2), (\omega_7, 1), (\omega_8, 0), (\omega_9, 0)\}$
2. $U\_V = \{(U_1, 12112), (U_2, 34112), (U_3, 12112), (U_4, 34112), (U_5, 12222)\}$
3. $T\_V = \left\{ \begin{pmatrix} 12112 \\ \{7,5\} \end{pmatrix}, \begin{pmatrix} 34112 \\ \{5,2\} \end{pmatrix}, \begin{pmatrix} 12222 \\ \{2,8\} \end{pmatrix} \right\}$

7.4.5 Common Control Channel and Default Channel Selection

Although previous works [7.11, 7.24] have assumed a Common Control Channel (CCC) for communication of signaling messages, such an assumption is not justified especially in a dynamic CR environment where channel conditions vary with time. However, CCC is highly required in CRN [7.1] mainly for real-time communication as it facilitates rapid message exchange and also helps in attaining quick consensus for resumption of transmission in the target channel following handoff. One approach is to select the CCC as the first channel in the *GA_TCS* algorithm of the first SU, as this is the best channel in terms of idle/busy ratio. As the SC performs periodic update of the *GA_TCS*, the CCC gets dynamically updated with varied network conditions. The SC broadcasts CCC to all SUs before sending periodic TCS updates. Also, to reduce signaling overhead over the CCC, the SC communicates $\left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ on response to *l_pro* to individual SUs and broadcasts the $\left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ information to all SUs during periodic updates using their operating frequencies. In such a case, SC continues to retransmit the $\left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ message to each SU until it receives an Acknowledgement from each SU. This is necessary in order to ensure that the SU transceiver is in receiving mode atleast once when SC transmits the information.

Additionally, whenever a new SU is admitted in the CRN, it must be provided with a default channel to initiate transmissions. One approach consistent with the proposed model in this chapter is that the SU sequentially senses the channels (starting from the lowest frequency) and on finding any idle channel after performing adaptive sensing (t_{ls}, t_{hs}), it waits for the broadcasted CCC information as periodically transmitted by the SC. Thereafter, the SU uses the CCC to send registration and subsequent channel allocation request to the SC. It must be noted that providing a default channel on response by SC to the SU depends on three critical factors namely; i) Call Admission Policy (balance between dropping and blocking strategies), ii) Fairness Policy (Preemption of NRT users for RT SUs), and iii) Queuing Model (Place RT users in high priority queues, etc.). This is beyond the scope of this current work and can be studied in future works.

7.4.6 Consensus among the SC and the SUs

This section describes all the different scenarios that arise with respect to ongoing SU transmissions and communication with the SC. While Scenario 1 describes the normal VoIP communication among the SUs, Scenarios 2 and 3 discuss the handoff operations by SUs while they reside in different and same PU transmission zones.

(i) ***<u>Scenario 1: Normal SU Transmissions</u>***

Let U_1 and U_2 be the caller and the callee involved in VoIP communication and they reside in the same zone under a common SC. The call flow for normal SU transmission follows the following steps.

1. U_1 gets a default channel for initiating transmission using CCC and CAC mechanisms.
2. Based on the session id as obtained from the call signaling information, if U_2 which is listening in some another idle channel, resides in the same zone, the SC provides the same default channel to U_2 for establishing communication with U_1.
3. U_1 and U_2 perform communication using RTS-CTS mechanism and initiate the underlying call signaling protocol (SIP or H.323) for establishing VoIP call.

(ii) <u>Scenario 2: SU handoff operation under different PU transmission zones.</u>

In this scenario, both U_1 and U_2 are under the jurisdiction of the same SC. However, they reside in different zones with respect to PU transmission. This implies that while the channel is sensed busy by U_1, the other SU (U_2) finds the same channel to be idle in a different zone.

1. U_1 invokes the *Three-level dropping decision* policy during ongoing communication.
2. After dropping from the current channel, U_1 selects one target channel either in the *Proactive* phase or in the *Reactive* phase.
3. U_2 on not receiving any RTP/RTCP information in the current channel waits for a certain response time (as denoted by t_{res}) and thereafter orients itself to the CCC for further information from SC. It keeps waiting for a maximum period of 2 X $T_{handoff}$ limit, after which the call is finally dropped.
4. As U_1 successfully receives a channel for transmission, it notifies SC using the *l_pro* message in the CCC. SC, in turn, notifies the same to the listening U_2 on the CCC.
5. U_1 and U_2 exchange RTS-CTS messages and resume communication in the new channel successfully.

(iii) Scenario 3: SU handoff operation under same PU transmission zone.

In this scenario, both U_1 and U_2 are under the jurisdiction of the same SC and reside in the same transmission zone of the PUs. This means channel conditions are identical for both the SUs.

1. Both U_1 and U_2 invoke the *Three-level dropping decision* policies almost simultaneously with a high probability as channel conditions are identical for both SUs.
2. SC has already updated the same $\left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ for both the SUs belonging to the same session.
3. Both SUs independently scan each channel in TCS for possible transmission. When any one SU selects any suitable channel to resume call, it sends RTS message twice, each after a span of t_{res} period. The other SU with all probability will also select the same idle channel as the channel conditions are identical and will receive atleast one incoming RTS message. It responds with CTS message and communication is established.
4. In the worst case when all channels in $\left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ are sensed busy, one SU (termed as the *Initiator* SU) first sends the *l_react* message to SC. Accordingly, it receives the Reactive TCS from SC, selects one suitable channel for transmission from this list and informs the SC of its current transmission channel as per the proposed design methodology. SC on receiving another *l_react* message from the other SU (*Follower* SU) of the same session id waits for incoming *l_pro* message from the *Initiator* SU and thereafter forwards the channel information to the *Follower* SU.
5. *Initiator* SU will continue to send RTS twice, one after every t_{res} period. *Follower* SU responds with CTS message on receiving any incoming RTS message and resumes communication.

It should be noted that the scenario where two communicating SUs are under the supervision of different SCs is not explicitly mentioned because in such a scenario, each SC can allot any idle channel as target channel to its subscribed SU following the proposed design model in this chapter and thereafter communicate with the other SC for transmission.

7.5 Mathematical Formulation and Analysis

In this section, the merits of the proposed algorithms are realized through mathematical modeling of critical system parameters based on which significant metrics such as dropping and handoff probabilities, handoff delay, effective transmission time, idle channel selection probability, etc. are derived and analyzed. As both SUs and the SCs work in close coordination, an analytical study is crucial towards determining these metrics from the systems perspective for validating the credibility of this work. In addition, the outcome of this section will further set the tone for comparative performance evaluation of the proposed methodology with earlier works in the next section.

7.5.1 Mathematical Analysis of Spectrum Handoff Algorithm: *ProReact* Segment

A channel ω in a CRN can be busy either when it is occupied by the PU or by the SU. ω witnesses intermittent on-off activity from PU [7.25]. Also being a candidate channel for SU allocation, ω is also allotted to other SUs by SC. Therefore, considering an earlier time instant t'' where $t'' < t'$, there is high probability that ω is either occupied by PU or by the SU. Let the average on-off time periods for SU and PU be denoted by λ_p, λ_s and μ_p , μ_s respectively. Therefore, probability that channel ω was busy at time t'' is given by,

$$\Pr_{busy(\omega)}^{(t'')} = \left(\Pr_{busy(\omega)}^{(t'')}\right)_{PU} + \left(\Pr_{busy(\omega)}^{(t'')}\right)_{SU} \tag{7.9}$$

Using Renewal theory [7.25], the probability that ω was occupied by PU at time t'' and also at time t is given by,

$$\left(\Pr_{busy(\omega)}^{(t/t'')}\right)_{PU} = \frac{\lambda p}{\lambda p + \mu p} + \frac{\mu p}{\lambda p + \mu p} e^{-\left(\frac{1}{\lambda p} + \frac{1}{\mu p}\right)(t - t'')} \tag{7.10}$$

Similarly for SU, we have

$$\left(\Pr_{busy(\omega)}^{(t/t'')}\right)_{SU} = \frac{\lambda s}{\lambda s + \mu s} + \frac{\mu s}{\lambda s + \mu s} e^{-\left(\frac{1}{\lambda s} + \frac{1}{\mu s}\right)(t - t'')} \tag{7.11}$$

Therefore, combining (7.10) and (7.11), the probability that channel ω is busy at time t is given by,

$$\Pr_{busy}^{(t)}(\omega) = \sum_{\substack{\lambda = \lambda s, \lambda p \\ \mu = \mu s, \mu p}} \left\{ \frac{\lambda}{\lambda + \mu} + \frac{\mu}{\lambda + \mu} e^{-\left(\frac{1}{\lambda} + \frac{1}{\mu}\right)(t - t'')} \right\} \tag{7.12}$$

In the *Proactive* phase, let $\Pr_{idle\_pro}^{(t+\Delta t)}[\omega]$ be the probability that a channel ω, where $\omega \in \left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ (that is, the proactive TCS) is idle when the U_i^{th} SU senses it at time interval $(t+\Delta t)$. The corresponding expression is derived as,

$$\Pr_{idle\_pro}^{(t+\Delta t)} = \left(\prod_{k=1}^{\omega-1} \Pr_{busy(k)}^{(t)}\right)\left(1 - \Pr_{busy(\omega)}^{(t)}\right) \tag{7.13}$$

Let $\left|\left(l_{pro}^{(t)}[\ ]\right)_{Ui}\right| = n_{pro}$. As per the algorithm, when all channels in n_{pro} are sensed busy, SU enters the *Reactive* phase. The different timing notations in this regard are defined in Table 7.3

Table 7.3 Different timing notations as used in the Analytical Model

Δt_1	total time incurred in sensing all the channels in Proactive TCS
Δt_2	time spent for receipt of Reactive TCS from SC
t_{s_i}	initial channel sensing time as per the algorithm
t_c	channel consensus time as per the proposed design
t_{sw}	channel switching time
$T_{max\_handoff}$	maximum handoff delay limit

The total number of channels in Reactive TCS that can be sensed by the SU within $T_{max\_handoff}$ limit is given by,

$$n_{maxsearch} = \frac{T_{max\_handoff} - (\Delta t_1 + \Delta t_2)}{t_{sw} + t_c + t_{s\_i}} \tag{7.14}$$

Let the total number of channels to be searched reactively at time t is given by, $\left(l_{react}^{(t)}[\ ]\right)_{Ui}$ where $\left|\left(l_{react}^{(t)}[\ ]\right)_{Ui}\right| = n_{maxsearch}$. The probability that atleast one channel is idle in $\left(l_{react}^{(t)}[\ ]\right)_{Ui}$ is given by,

$$\Pr_{idle\_react}^{(t)} = \sum_{\substack{k=1 \\ v\in A, v'\notin A, \forall A \subseteq \left(l_{react}^{(t)}[\ ]\right)}}^{n_{max\_search}} \left\{ \prod \left(1 - \Pr_{busy}{}_{(v)}^{(t)}\right) \prod \Pr_{busy}{}_{(v')}^{(t)} \right\} \tag{7.15}$$

where $\Pr_{busy_{(v)}}^{(t)}$ = busy probability for channel v at time t. The expression for $\Pr_{busy_{(\omega)}}^{(t)}$ has already been obtained in (7.12).

Given n_{idle} number of idle channels, the probability of selecting one channel randomly and uniformly is 1/ n_{idle}. Therefore, combining (7.12), (7.14) and (7.15), the probability of selecting a particular idle channel ω in the *Reactive* phase of *ProReact* segment is given by (7.16).

$$\Pr_{idle\_react}^{(t)} = \left(1 - \Pr_{busy_{(\omega)}}^{(t)}\right) \sum_{\substack{k=1 \\ \forall A \subseteq \left(l_{react}^{(t)}[\ ]\right)/\{\omega\}}}^{\left(\frac{T_{max\_handoff} - (\Delta t_1 + \Delta t_2)}{t_{sw} + t_c + t_{s\_i}}\right)} \left\{ \frac{1}{|A|+1} \prod_{v\in A} \left(1 - \Pr_{busy_{(v)}}^{(t)}\right) \prod_{\substack{v'\notin A, \\ v'\notin\{\omega\}}} \Pr_{busy_{(v')}}^{(t)} \right\} \tag{7.16}$$

This leads to the following Proposition.

Proposition 7.1

The probability of finding atleast a single idle channel during the two-phase spectrum handoff strategy increases with higher values of n_{pro} and $n_{maxsearch}$, but at the price of higher handoff delay.

Proof: Let $\left(\Pr_{busy\_total}^{(t)}\right)_{Ui}$ be the probability that all channels sensed by the SU (U_i) are busy, which will eventually lead to call drop and is given by,

$$\left(\Pr_{busy\_total}^{(t)}\right)_{Ui} = \left(\Pr_{busy\_pro}^{(t1)}\right)_{Ui} \times \left(\Pr_{busy\_react}^{(t2)}\right)_{Ui} \tag{7.17}$$

where $\left(\Pr_{busy\_pro}^{(t1)}\right)_{Ui}$, $\left(\Pr_{busy\_react}^{(t2)}\right)_{Ui}$ denote that all channels in n_{pro} and

$n_{maxsearch}$ respectively are detected busy, and $t > t_1 > t_2$. Deriving expressions for these two probabilities,

Case 1: *Proactive* phase

Timeline	Channel Busy Probability
t_0	$\Pr_{busy(\omega_1)}^{(t_0)}$
$t_0 + \Delta t_p$	$\Pr_{busy(\omega_2)}^{(t_0+\Delta t_p)}$
⋮	⋮
$t_0 + n_{pro}\Delta t_p$	$\Pr_{busy(\omega_{npro})}^{(t_0+n_{pro}\Delta t_p)}$

$$\Rightarrow \left(\Pr_{busy\_pro}^{(t_1)}\right)_{Ui} = \prod_{k=1}^{n_{pro}}\left(\Pr_{busy(\omega_k)}^{(t_0+k\Delta t_p)}\right)_{Ui} \tag{7.18}$$

where $\omega_k \in \left(l_{pro}^{(t_0)}[\ ]\right)_{Ui}$ and

$t_1 = t_0 + n_{pro}\Delta t_p$

Case 2: *Reactive* Phase

Timeline	Channel Busy Probability
t_n	$\Pr_{busy\left(\omega_{n_{pro}+1}\right)}^{(t_n)}$
$t_n + \Delta t_r$	$\Pr_{busy\left(\omega_{n_{pro}+2}\right)}^{(t_n+\Delta t_r)}$
⋮	⋮
$t_n + n_{maxsearch}\Delta t_r$	$\Pr_{busy\left(\omega_{n_{pro}+n_{maxsearch}}\right)}^{(t_n+n_{react}\Delta t_p)}$

$$\Rightarrow \left(\Pr_{busy\_react}^{(t_2)}\right)_{Ui} = \prod_{k=1}^{n_{maxsearch}}\left(\Pr_{busy\left(\omega_{n_{pro}+k}\right)}^{(t_n+k\Delta t_r)}\right) \tag{7.19}$$

where $\omega_k \in \left(l_{react}^{(t_n)}[\ ]\right)_{Ui}$ and

$t_n = t_0 + n_{pro}\Delta t_p + t_c$,

$t_2 = t_n + n_{maxsearch}\Delta t_r$

Therefore, from (7.17),

$$\left(\Pr_{busy\_total}^{(t)}\right)_{Ui} = \prod_{k=1}^{n_{pro}}\left(\Pr_{busy(\omega_k)}^{(t_0+k\Delta t_p)}\right)_{Ui} \times \prod_{k=1}^{n_{maxsearch}}\left(\Pr_{busy\left(\omega_{n_{pro}+k}\right)}^{(t_n+k\Delta t_r)}\right)_{Ui}$$

$$= \prod_{k=1}^{n_{pro}} \left\{ 1 - \left(\Pr_{idle\_pro}^{\left(t_0 + k\Delta t_p\right)}(\omega_k) \right)_{Ui} \right\} \times \prod_{k=1}^{n_{maxsearch}} \left\{ 1 - \left(\Pr_{idle\_react}^{\left(t_n + k\Delta t_r\right)}\left(\omega_{n_{pro}+k}\right) \right)_{Ui} \right\} \quad (7.20)$$

It is therefore evident from (7.20) that as both n_{pro} and $n_{maxsearch}$ increase, $\left(\Pr_{busy\_total}^{(t)}\right)_{Ui}$ decreases, which implies that the probability of finding atleast a single idle channel as given by, 1- $\left(\Pr_{busy\_total}^{(t)}\right)_{Ui}$ increases. However, increasing the value of n_{pro} and $n_{maxsearch}$ also implies higher handoff delay limits with respect to $T_{handoff}$ and $T_{max\_handoff}$ respectively. Thus, ensuring an idle channel for SU transmission with higher probability also effectively increases the overall handoff delay.

(i) ***Calculation of Channel Busy Probability from SU perspective***

Let SU senses the currently occupied channel for n_p number of time slots where p is equal to single sensing slot duration. Therefore, duration of the ith sensing cycle in channel ω is $ts_{(\omega)}^{(i)} = n_p \times p$. The channel ω is considered busy by the SU transceiver if the received power (Pow_{rec}) is greater than a certain threshold power (Pow_{th}). Hence, in one slot duration of p, $e_{th} = Pow_{th} \times p$ and $e_{rec} = Pow_{rec} \times p$, where e_{th} and e_{rec} are the threshold and detected energy from the received power. If $e_{rec} \geq e_{th}$, consider a binary variable B_i=1 for the ith slot of duration p in n_p. The channel is considered busy when $\sum_{i=1}^{np} Bi \geq nth$,where n_{th} is the threshold value. Therefore, probability that the channel is sensed busy by Ui th SU at time t is expressed as,

$$\left(\Pr_{s\_busy(\omega)}^{(t)}\right)_{Ui} = \Pr\left\{ \sum_{j=1}^{np} \left(\Pr\left(Powrec \times p \geq eth\right)\right)_{Ui} \geq nth \right\} \quad (7.21)$$

Selection of proper n_p, p and n_{th} is crucial to properly estimate the channel busy probability $\left(\Pr_{s\_busy\omega}^{(t)}\right)_{Ui}$ as explained in the following proposition.

Proposition 7.2

Considering imperfect channel sensing by SU, the channel busy probability as sensed by SU in one CR cycle is primarily dependent on the values of n_p and n_{th} and significantly contributes to false-alarm and miss-detection cases.

Proof: Probability that a channel is sensed busy by SU= Probability that the channel is detected busy when it is actually busy + Probability that the channel is sensed busy when it is actually idle.

This implies from (7.21),

$$\Pr\left\{\sum_{j=1}^{np}\left(\Pr\left(Pow_{rec}\times p\geq eth\right)\right)_{Ui}\geq nth\right\}=\left\{\left(1-P_{md}^{(\omega)}\right)+P_{FA}^{(\omega)}\right\} \qquad (7.22)$$

where P_{md}, P_{FA} = probability of miss-detection [7.22] and false alarm [7.22] respectively. Two cases arise with respect to (7.22) as discussed below.

i) Keeping n_p fixed, when n_{th} is increased in (7.22), $\left(\Pr_{s\_busy\omega}^{(t)}\right)_{Ui}$ decreases. This can be logically explained as follows. When the threshold energy for PU detection is increased, the probability of the received energy being greater than the threshold energy decreases, thereby reducing the value of $\left(\Pr_{s\_busy\omega}^{(t)}\right)_{Ui}$. This also reduces P_{FA} but at the cost of higher P_{md}.

ii) Keeping n_{th} fixed, as n_p increases in (7.22), $\left(\Pr_{s\_busy\omega}^{(t)}\right)_{Ui}$ increases. This is because with increase in the number of sensing slots per cycle while keeping the threshold value constant, the false alarm probability P_{FA} increases significantly, even though P_{md} decreases.

7.5.2 Mathematical Analysis of Spectrum Handoff Algorithm: *VAST* Segment

After the *ProReact* segment, now the *VAST* policy is mathematically analyzed. The most significant metric is the probability to drop from the current channel which is derived in all the three dropping decision phases. Let n_{ls} and

n_{hs} correspond to the total number of sensing slots for t_{ls} and t_{hs} respectively, and their corresponding threshold be denoted by n_{th_ls} and n_{th_hs} respectively.

(i) Level – 1 (Cons_sense Dropping Policy)

Probability to drop from the current channel ω due to *Cons_sense* at the *i*th sensing slot = Probability that ω is sensed busy at (i-2) sensing slot during t_{ls} period X Probability that ω is sensed busy at (i-1) and *i*th sensing slot during t_{hs} period. (7.23)

Let $\left(\Pr_{drop\_cons\_sense(\omega)}^{(ts_i)}\right)_{Ui}$ be the probability that U_i th SU drops due to *Cons_sense* from channel ω at the end of *i*th sensing slot (ts_i) and is expressed from (7.23) as follows.

$$\left(\Pr_{drop\_cons\_sense(\omega)}^{(ts_i)}\right)_{Ui} = \Pr\left\{\sum_{j=1}^{nls}\left(\Pr\left(Powrec(j)\times p \geq eth\right)\right)_{Ui} \geq nth\_ls\right\}_{i-2} \times \prod_{k=i-1,i} \Pr\left\{\sum_{j=1}^{nhs}\left(\Pr\left(Powrec(j)\times p \geq eth\right)\right)_{Ui} \geq nth\_hs\right\}_{k} \quad (7.24)$$

(ii) Level – 2 (Cons_tx Dropping Policy)

Probability to drop from the current channel ω due to *Cons_tx* at the *i*th sensing slot = Probability that ω is sensed busy at (i-4) sensing slot during t_{ls} period X Probability that ω is sensed busy at (i-2) and *i*th sensing slot during t_{ls} period. (7.25)

Let $\left(\Pr_{drop\_cons\_tx(\omega)}^{(ts_i)}\right)_{Ui}$ be the probability that U_i th SU drops due to *Cons_tx* from channel ω at the end of *i*th sensing slot (ts_i) and *j*th transmission slot (td_j)and is expressed from (7.25) as,

$$\left(\Pr_{drop\_cons\_tx(\omega)}^{(ts_i,td_j)}\right)_{Ui} = \prod_{k=i-4,i-2,i} \Pr\left\{\sum_{j=1}^{nls}\left(\Pr\left(Powrec(j)\times p \geq eth\right)\right)_{Ui} \geq nth\_ls\right\}_{k} \quad (7.26)$$

(iii) ***Level – 3 (Alt_tx Dropping Policy)***

With respect to the *Alt_tx* policy, two scenarios are considered with respect to both SU and system perspective.

Scenario 1: Probability of entering $t_{hd}(j)$ at the end of $t_{ls}(i)$ = Probability that channel remains idle during $t_{ls}(i)$

$$= \Pr\left\{\sum_{j=1}^{nls}\left(\Pr\left(Powrec\left(j\right)\times p \geq eth\right)\right)_{Ui} < nth\_ls\right\}_i \tag{7.27}$$

Case 2: Probability of entering $t_{ld}(j)$ at the end of $t_{hs}(i)$ = Probability that channel remains busy during $t_{ls}(i\text{-}1)$ slot X Probability that channel remains idle during $t_{hs}(i)$ period

$$= \Pr\left\{\sum_{j=1}^{nls}\left(\Pr\left(Powrec\left(j\right)\times p \geq eth\right)\right)_{Ui} \geq nth\_ls\right\}_{i-1} \times \Pr\left\{\sum_{j=1}^{nhs}\left(\Pr\left(Powrec\left(j\right)\times p \geq eth\right)\right)_{Ui} < nth\_hs\right\}_i \tag{7.28}$$

Combining these two cases as per the proposed algorithm, the probability that U_ith SU drops due to *Alt_tx* decision from channel ω at the end of ith sensing slot (ts_i) and jth transmission slot (td_j) is expressed as follows.

$$\left(\Pr_{drop\_alt\_tx(\omega)}^{(ts_i,td_j)}\right)_{Ui} = \Pr\left[\sum_{m=1}^{ncycle}\left[\Pr\left\{\sum_{j=1}^{nls}\left(\Pr\left(Powrec\left(j\right)\times p \geq eth\right)\right)_{Ui} \geq nth\_ls\right\}_{i-1} \times \Pr\left\{\sum_{j=1}^{nhs}\left(\Pr\left(Powrec\left(j\right)\times p \geq eth\right)\right)_{Ui} < nth\_hs\right\}_i\right]_m \geq 1/2\sum_{m=1}^{ncycle}\left[\Pr\left\{\sum_{j=1}^{nls}\left(\Pr\left(Powrec\left(j\right)\times p \geq eth\right)\right)_{Ui} < nth\_ls\right\}\right]_m\right] \tag{7.29}$$

Finally, the overall probability to drop from the current channel following the *Three-level dropping decision* policy can be expressed as,

$$\left(\Pr_{drop\_current(\omega)}^{(ts_i,td_j)}\right)_{Ui} = \left(\Pr_{drop\_cons\_sense(\omega)}^{(ts_i)}\right)_{Ui} + \left(\Pr_{drop\_cons\_tx(\omega)}^{(ts_i,td_j)}\right)_{Ui} + \left(\Pr_{drop\_alt\_tx(\omega)}^{(ts_i,td_j)}\right)_{Ui} \tag{7.30}$$

It must be noted that these three dropping decision events are mutually exclusive but not exhaustive.

7.5.3 Derivation of Important System Metrics

Based on the underlying mathematical framework as already explained, important system metrics concerning VoIP and CRN are defined and derived for performance evaluation in subsequent sections.

(i) ***Handoff Probability***

Handoff probability in this chapter is defined as the probability of obtaining atleast one single idle channel for resuming communication after successfully vacating the current channel of operation and is expressed as follows.

$\left(\Pr_{handoff}^{(t')}\right)_{Ui}$ = Probability that SU drops from the current channel X Probability of obtaining atleast one idle channel successfully

$$= \left(\Pr_{drop\_current(\omega)}^{(ts_i, td_j)}\right)_{Ui} \times \left(1 - \Pr_{busy\_total}^{(t'')}\right)_{Ui},\ t' = t(i,j) + t'' \qquad (7.31)$$

where t''= total time incurred during handoff, $t(i,j)$ = elapsed time after ith sensing (ts_i) and jth transmission (td_j) time intervals. $\left(\Pr_{drop\_current(\omega)}^{(ts_i, td_j)}\right)_{Ui}$ and $\left(\Pr_{busy\_total}^{(t'')}\right)_{Ui}$ are derived in (7.30) and (7.17) respectively.

(ii) ***Dropping Probability***

Dropping probability is defined as the probability that a particular SU is dropped from the CRN due to lack of availability of an idle channel resulting in subsequent termination of the VoIP communication and is given by,

$\left(\Pr_{drop}^{(t')}\right)_{Ui}$ = Probability that SU drops from the current channel X Probability that all channels sensed by SU are busy

$$= \left(\Pr_{drop\_current(\omega)}^{(ts_i, td_j)}\right)_{Ui} \times \left(\Pr_{busy\_total}^{(t'')}\right)_{Ui},\ t' = t(i,j) + t'' \qquad (7.32)$$

where all the symbols have their usual meanings as already defined.

(iii) Average Number of Channels Sensed before Successful handoff

The average number of channels sensed by SU before finally selecting an idle channel for resuming VoIP call directly influences the handoff delay experienced after every instance of channel drop and is a critical metric for determining the success of any handoff algorithm. Let N be the maximum number of channels that can be sensed by U_i th SU and is given by,

$$N = n_{pro} + n_{react} \text{ where } t_n = t_0 + n_{pro}\Delta t_p + t_c \tag{7.33}$$

where all symbols have their usual notations as per Table 7.3.

Let handoff occurs at the kth channel. Given that the spectrum handoff is successful, the probability that the handoff occurs after sensing k number of channels at time t_2 is given by the following expression.

$$\left(\Pr\left(h = k / h \le N\right)\right)_{Ui}^{t2} = \frac{\Pr\left(h = k \cap h \le N\right)}{\Pr\left(h \le N\right)}$$

$$= \frac{\textit{Probability that handoff occurs at kth trial}}{\textit{Probability that handoff is successful}} \tag{7.34}$$

Case 1: *Proactive* Phase

Probability that idle channel selected at kth trial	=	Probability that (k-1) channels are sensed busy	X	Probability that kth channel is idle and suitable for selection

$$\Rightarrow \Pr{}_{idle\_pro}^{(t2)}\left(h = k\right) = \prod_{l=1}^{k-1}\left(1 - \Pr{}_{idle\_pro}^{(t1 + l\Delta t)}\left[\omega_l\right]\right) \times \Pr{}_{idle\_pro}^{(t2)}\left[\omega_k\right] \tag{7.35}$$

where t_1 = time corresponding to sensing first channel, Δt = time incurred in sensing each channel in n_{pro}, $t_2 = t_1 + k\Delta t$, $k <= n_{pro}$.

Case 2: *Reactive* Phase

Probability that idle channel selected at kth trial	=	Probability that (k-1) channels are sensed busy	X	Probability that kth channel in $\left(l_{pro}^{(t0)}[\ ]\right)_{Ui}$ is idle and suitable for selection

= Probability that all channels in $\left(l_{pro}^{(t)}[\ ]\right)_{Ui}$ are sensed busy x Probability that $(k\text{-}n_{pro})$ no. of channels in $\left(l_{react}^{(t)}[\ ]\right)_{Ui}$ are sensed busy

x Probability that kth channel $\left(l_{react}^{(t_n)}[\ ]\right)_{Ui}$ is idle and suitable for selection

$$\Rightarrow \Pr_{idle\_react}^{(t_n)}(h=k)=\prod_{l=1}^{n\,pro}\left(1-\Pr_{idle\_pro}^{(t_1+l\Delta t)}[\omega l]\right)\times$$

$$\prod_{l=n\,pro+1}^{k-1}\left(1-\Pr_{idle\_react}^{(t_n+l\Delta t')}[\omega l]\right)\times \Pr_{idle\_react}^{(t_2)}[\omega k] \tag{7.36}$$

where Δt′= time spent in sensing each channel in $n_{maxsearch}$, $t_2 = t1+n_{pro}\Delta t+ k\Delta t'$, $k <= n_{maxsearch}$.

Finally, the probability that handoff is successful at time t_2 is given by $\left(\Pr_{handoff}^{(t_2)}\right)_{Ui}$, which has been obtained in (7.31).

Therefore from (7.34), the expected number of channels sensed before successful handoff is given by,

$$E(h/h\le N)_{Ui}^{t2}=\sum_{k=2}^{N}\left\{k\times\left(\Pr(h=k/h\le N)\right)_{Ui}^{t2}\right\}$$

$$=\sum_{k=2}^{N}\left[\frac{k}{\left(\Pr_{handoff}^{(t_2)}\right)_{Ui}}\times H_0\left\{\prod_{l=1}^{k-1}\left(1-\Pr_{idle\_pro}^{(t_1+l\Delta t)}[\omega_l]\right)\times\Pr_{idle\_pro}^{(t_2)}[\omega_k]\right\}+\right.$$

$$\left.H_1\left\{\prod_{l=1}^{n_{pro}}\left(1-\Pr_{idle\_pro}^{(t_1+l\Delta t)}[\omega_l]\right)\times\prod_{l=n_{pro}+1}^{k-1}\left(1-\Pr_{idle\_react}^{(t_n+l\Delta t')}[\omega_l]\right)\times\Pr_{idle\_react}^{(t_2)}[\omega_k]\right\}\right] \tag{7.37}$$

where H_0=1 when $k<=n_{pro}$, 0 otherwise, H_1=1 when $k>n_{pro}$, 0 otherwise. It must be noted that k=1 implies that the SU selects the first idle channel in the TCS with the probability $\Pr_{idle\_pro}^{(t_0)}[\omega\, 1]$.

(iv) ***Handoff Delay***

Handoff delay is defined as the total time incurred in resuming transmission in a newly selected idle channel following successful spectrum handoff after halting communication and subsequent dropping from the current channel in operation and is expressed as,

$$\left(D_{handoff}\right)_{Ui} = \Pr_{idle\_pro}^{(t_0)}\left[\omega_1\right]\left(t_{s\_i} + t_c + t_{sw}\right) \quad \forall k = 1$$

$$= \sum_{k=2}^{N}\left[\frac{k}{\left(\Pr_{handoff}^{(t_2)}\right)_{Ui}} \times H_0\left\{\prod_{l=1}^{k-1}\left(1-\Pr_{idle\_pro}^{(t_1+l\Delta t)}\left[\omega_l\right]\right)\times \Pr_{idle\_pro}^{(t_2)}\left[\omega_k\right]\right\} + \right.$$

$$\left. H_1\left\{\prod_{l=1}^{n_{pro}}\left(1-\Pr_{idle\_pro}^{(t_1+l\Delta t)}\left[\omega_l\right]\right)\times \prod_{l=n_{pro}+1}^{k-1}\left(1-\Pr_{idle\_react}^{(t_n+l\Delta t')}\left[\omega_l\right]\right)\times \Pr_{idle\_react}^{(t_2)}\left[\omega_k\right]\right\}\right]$$

$$\times t_{s\_i} + t_c + t_{sw} \quad \forall k \geq 2$$

$$(7.38)$$

where all symbols have their usual meanings as already defined. A critical aspect that is worth mentioning is that since $\left(l_{pro}^{(t)}[\;]\right)_{Ui}$ gets updated at regular intervals by the SC, this work does not deal with the total number of handoffs done or the cumulative handoff delay, as studied in earlier works [7.6]. Rather, dealing with real-time calls, this chapter attaches maximum importance to the average number of channels searched before every successful spectrum handoff and the corresponding handoff delay, both of which have been derived in (7.37) and (7.38) respectively.

(v) ***Average Transmission Time available to the SU***

The mean transmission time for SU at the end of *i*th sensing and *j*th transmission time slot at a particular channel *ω* can be expressed as,

$$\left(Tx(i,j)\right)_{Um} = Probability\ of\ tx.\ in\ the\ channel\ \omega \times$$

$$Average\ tx.\ duration\ in\ \omega\ till\ (i,j) \qquad (7.39)$$

From the SU perspective, we have the following scenario.

$$(Tx(i,j))_{Um}^{SU} = \left\{1-\left(\Pr_{drop\_current\ (\omega)}^{(ts_i, td_j)}\right)_{Um}\right\} \times$$

$$\sum_{till\ (i,j)} \left\{\left(1-\left(\Pr_{s\_busy(\omega)}^{(ts=tls(i))}\right)_{Um}\right) thd + \left(\Pr_{s\_busy(\omega)}^{(ts=tls(i-1))}\right)_{Um} \left(1-\left(\Pr_{s\_busy(\omega)}^{(ts=ths(i))}\right)_{Um}\right) tld(j)\right\}$$

(7.40)

From the system perspective, considering the effects of false alarm and miss-detection, interference occurs when both SU and PU transmit, resulting in non-effective communication duration. Two scenarios arise as explained below.

Scenario 1: Interference during transmission at t_{hd} duration

Two cases occur when miss-detection occurs at the current channel.

Case 1: The channel does not change state for the entire t_{hd} duration. Thus miss-detection occurs for the entire duration.

Case 2: Miss-detection occurs only till the channel changes state within the t_{hd} duration.

Therefore, based on [7.22] and as per our algorithm,

$$(t_{\inf\_thd}(j))_{Um}^{\omega} = \frac{1}{P_{md}^{(\omega)} \Pr_{busy(\omega)}^{(t=t_{hd}(j))} + \left(1-P_{FA}^{(\omega)}\right)\left(1-\Pr_{busy(\omega)}^{(t=t_{hd}(j))}\right)} \left[\left[\frac{1}{\alpha}\left(1-e^{-\alpha t_{hd}(j)}\right) \times P_{md}^{(\omega)} \times \Pr_{busy(\omega)}^{(t=t_{hd}(j))}\right] + \right.$$

$$\left.\left[\left\{t_{hd}(j) - \frac{1}{\beta}\left(1-e^{-\beta t_{hd}(j)}\right)\right\} \times \left(1-P_{FA}^{(\omega)}\right)\left(1-\Pr_{busy\ (\omega)}^{(t=t_{hd}(j))}\right)\right]\right] \quad (7.41)$$

Scenario 2: Interference during transmission at t_{ld} duration

Unlike the previous scenario, the probability of miss-detection is very less here as we have two sensing durations (t_{ls}, t_{hs}) before entering t_{ld} as per the algorithm. Therefore, here we consider only the second case where miss-detection occurs only till the channel changes state within the t_{ld} duration. Hence proceeding as above,

$$(t_{\inf\_tld}(j))_{Um}^{\omega} = \frac{\left[\left\{tld(j) - \frac{1}{\beta}\left(1-e^{-\beta tld(j)}\right)\right\} \times \left(1-P_{FA}^{(\omega)}\right)\left(1-\Pr_{busy\ (\omega)}^{(t=tld(j))}\right)\right]}{P_{md}^{(\omega)} \Pr_{busy\ (\omega)}^{(t=tld(j))} + \left(1-P_{FA}^{(\omega)}\right)\left(1-\Pr_{busy\ (\omega)}^{(t=tld(j))}\right)}$$

(7.42)

Therefore, combining (7.41) and (7.42), the average transmission time for SU in a particular channel can be expressed as,

$$(Tx(i,j))_{Um}^{System} = \left\{1-\left(\Pr_{drop\_current(\omega)}^{(tsi,tdj)}\right)_{Um}\right\} \sum_{till\,(i,j)} \left[\left(1-\left(\Pr_{s\_busy(\omega)}^{(ts=tls(i))}\right)_{Um}\right)\left\{thd(j)-(t\inf\_thd(j))_{Um}^{\omega}\right\}\right.$$
$$\left. + \left(\Pr_{s\_busy(\omega)}^{(ts=tls(i-1))}\right)_{Um}\left(1-\left(\Pr_{s\_busy(\omega)}^{(ts=ths(i))}\right)_{Um}\right)\left\{tld(j)-(t\inf\_tld(j))_{Um}^{\omega}\right\}\right]$$

(7.43)

(vi) *Packet Loss Percentage for VoIP session*

VoIP communication occurs in the form of talkspurts [7.26]. Hence, the SU transmits during the talkspurt period and remains idle during the silent suppression period with probabilities P_{busy} and P_{idle} respectively. Accordingly, the VoIP traffic for a single SU is formulated as an on-off model with $1/\gamma$ and $1/\delta$ as on and off periods respectively, that are exponentially distributed [7.20]. Therefore, the probability with which the SU is busy transmitting is given by,

$$P_{busy}(V) = \frac{\gamma-1}{\gamma-1+\delta-1} \qquad (7.44)$$

Let $R\omega\ (V)$ be the VoIP transmission rate for a SU where ω denotes an idle channel and V represents the VoIP transmission. It is worth mentioning that the average VoIP transmission rate is 50 or 33.3 packets per second and depends on the codecs implemented [7.27]. Based on the mathematical framework, the average number of VoIP packets sent by the SU and packets that were successfully transmitted are given by (7.45) and (7.46) respectively.

$$\eta_{tx\_packets}^{SU} = (Tx(i,j))_{Um}^{SU} \times P_{busy}(V) \times R_{\omega}(V) \qquad (7.45)$$

$$\eta_{tx\_packets}^{overall} = (Tx(i,j))_{Um}^{System} \times P_{busy}(V) \times R_{\omega}(V) \qquad (7.46)$$

Therefore, the percentage of packets lost in (i, j) period is given by,

$$\eta\%_{loss} = \frac{\eta_{tx\_packets}^{SU} - \eta_{tx\_packets}^{System}}{\eta_{tx\_packets}^{SU}} \qquad (7.47)$$

The overall packet loss is greatly affected by this $\eta\%_{loss}$ and must be below 5-6% for QoS sensitive VoIP communication.

7.6 Performance Evaluation and Discussion

The proposed spectrum handoff algorithm along with channel allocation strategies are now evaluated for performance efficiency with respect to recorded works in literature. In this regard, a model for VoIP based CRN is designed in MATLAB. The total number of channels is 30. PUs follow an exponential on-off model, and the on-off rates vary randomly. Total handoff limit for VoIP SUs is kept at 2s, and the CR cycle parameters are defined as, t_{ls}=50 ms, t_{hs}=100 ms, t_{ld}=100 ms, and t_{hd}=250 ms. Further, t_{s_i}=25 ms, t_c =100ms, and t_{sw}=25 ms. Initially, comparative performance evaluation of the proposed *GA_TCS* algorithm is carried out under the simulation environment with respect to other approaches in Table 7.4. It is observed that *GA_TCS* selects the maximum number of channels with the highest cumulative idle time compared to other approaches. In doing so, the proposed TCS calculation incurs minimum algorithmic iterations among all other options. This establishes the superiority and applicability of GA_TCS algorithm for real-time VoIP communication.

Table 7.4 Performance Evaluation of *GA_TCS* Algorithm with existing approaches

Metrics	Int_DP	Int_GR_Desc	Int_GR_Asc	Int_GR_Ratio	Frac_GR_Desc	Frac_GR_Asc	GA_TCS Algorithm
Total Idle time (ms)	4942.5	479	4648.5	4755.5	1906	5148.5	5504.5
Average no. of Channels selected	6	1	7	6	4	8	8
Iterations	28000	40	40	40	40	40	40
Int_DP, Int_GR_Desc, Int_GR_Asc, Int_GR_Ratio = Integer Knapsack using Dynamic Programming, Greedy Algorithm with Descending value, Ascending weight and descending value/weight Ratios respectively. Frac_GR_Desc, Frac_GR_Asc = Fractional Knapsack using Greedy Algorithm with Descending value, Ascending weight respectively.							

In addition, based on previous works, three other cases are considered; i) *React* (Purely Reactive handoff [7.15]), ii) *Pro-1* (Proactive Handoff to channel whose idle time is more than busy period [7.5]), and iii) *Pro-2* (Variant of *Pro-1*:Proactive Handoff to channel whose idle time is atleast 1.5 times more than busy period). Since both the models for SC and SU are tightly coupled in this chapter, their performance is jointly analyzed using the notation *Algo*.

Initially, the channels do not witness drastic variation in their usage characteristics. This corresponds to the best and average case network scenarios. As observed from Fig. 7.7, *Algo* selects channels with higher idle periods as compared to *Pro-1* and *React* during the *Proactive* and *Reactive* phases respectively. The problem with *Pro* and *Pro-1* is that both of them select a limited number of channels in the TCS. On the other hand, *React* Algorithm selects channels on an on-demand random fashion, resulting in selection of channels with high busy probabilities. Our design model improves channel selection performance with respect to the algorithms by choosing maximum number of idle channels while maintaining comparatively low busy probabilities.

In the next phase, the channel conditions are varied with time, denoting a highly dynamic network corresponding to the worst case scenario. It is seen from Fig. 7.8 that *Pro-2* selects the best channels with lowest busy channel probabilities (0.53) while *Pro-1* and *React* algorithms select channels with absolute busy channel probability. *Algo* still performs better in this case, by selecting higher number of channels for handoff (compared to *Pro-2*) with channel busy probabilities less than 1(compared to *React* and *Pro-1*).

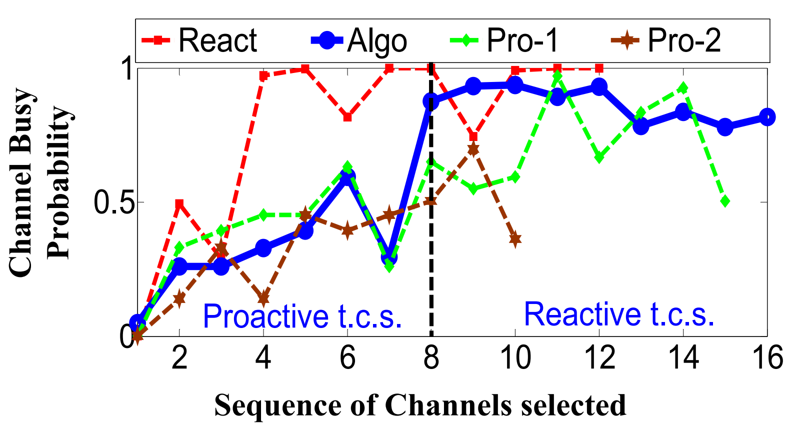


Fig. 7.7 Channel Busy Probability versus the sequence of target channels under the 4 algorithms with respect to the best/average case scenario

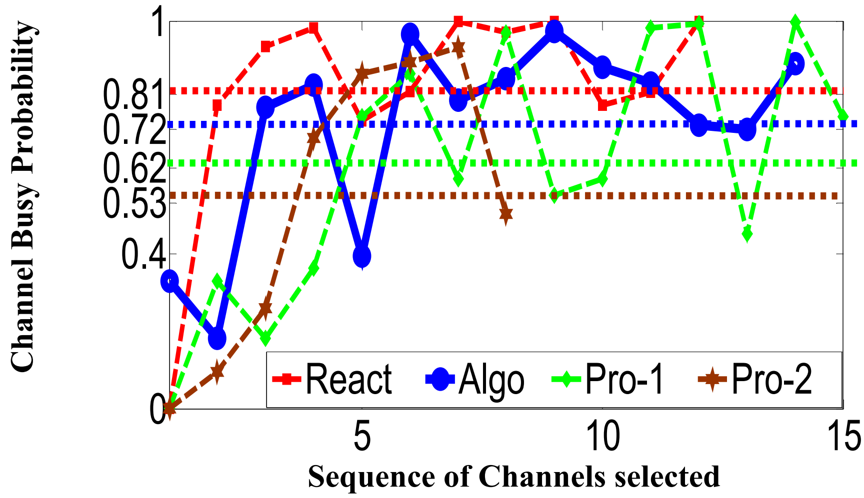


Fig. 7.8 Variation in the Channel Busy Probability with respect to the 4 algorithms in the worst case scenario

Better channel selection leads to a minimum 10% reduction in dropping probability p_{drop} with respect to the other scenarios, and further provides maximum transmission duration for VoIP calls (180 ms in the best/average case and 50ms in the worst case scenario) as seen in Fig. 7.9. While p_{drop} reaches 1 in the 5th and 7th time slot for *React* and *Pro-1* respectively, and 0.99 for *Pro-2* at the end of 20th slot, it remains below 0.9 even at the end of the 20th slot for *Algo*, thereby proving its efficiency.

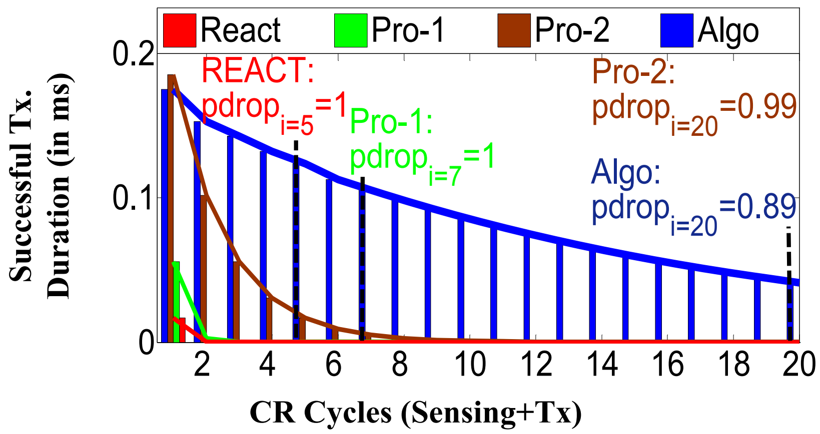


Fig. 7.9 Successful SU Transmission Duration with respect to the 4 algorithms

In addition, the handoff delay at the end of the 10th CR cycle slot is plotted in Fig. 7.10. It is seen that *Pro-2* and *React* experience the lowest and highest handoff delays respectively. On the contrary, *Algo* performs adaptively, as it ensures handoff delay below 150 ms during the average-case *Proactive* phase (40% lower than *Pro-1* and *Pro-2*) and 600 ms in the worst-case *Reactive* phase (60% lower than *React* Scenario).

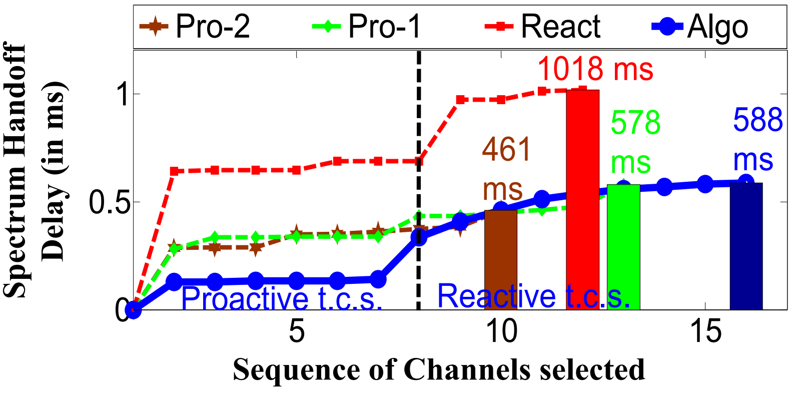


Fig. 7.10 Average Spectrum Handoff Delay with respect to the different channels accessed under the 4 scenarios

It is also evident from Fig. 7.10 that *Algo* selects the maximum number of channels for possible handoff, compared to the other algorithms.

Finally, the inter-working of the dropping decision and handoff operations in this chapter are analyzed in the designed simulation model. Fig.7.11 measures the number of dropping instances after applying the proposed algorithm. As observed from the figure, maximum number of dropping instances is recorded due to *Alt_tx* condition during the *Proactive* phase of the *ProReact* algorithm, as the channel conditions are better in the *Proactive* phase (corresponding to the best/average case scenarios). On the other hand, both *Cons_sense and Cons_tx* conditions result in higher number of dropping instances than *Alt_tx* in the *Reactive* phase of *ProReact* Algorithm, as the *Reactive* phase is the worst case scenario with high busy channel probabilities. It is also proved that with increase in the PU arrival probability, the number of dropping instances also increases.

Thus, it is inferred from simulation results that the developed algorithms in this chapter optimally select the channels with high idle probabilities while reducing both the channel dropping probability and handoff delay, and maximizing the transmission duration, and thereby successfully maintain the QoS of VoIP calls.

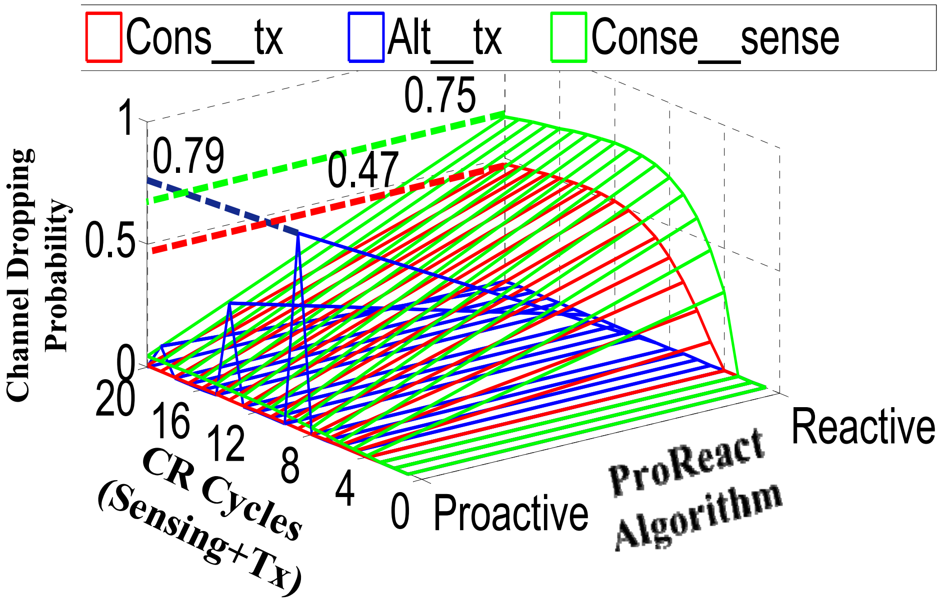


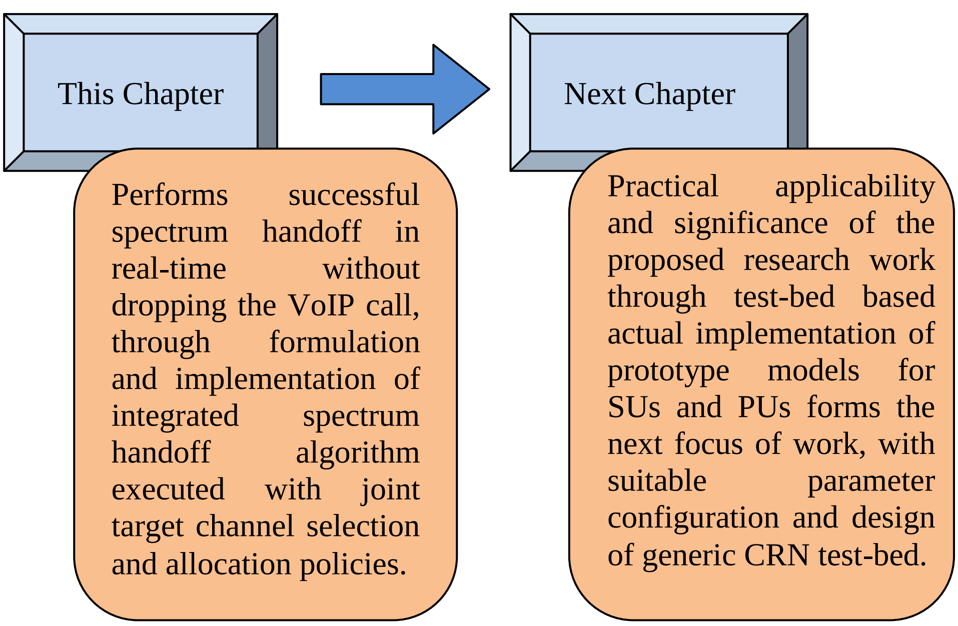
Fig. 7.11 Variation in Channel Dropping Probability with CR cycles under *Proactive* and *Reactive* phases of *ProReact* Module in the Proposed Algorithm

7.7 Summary

Considering spectrum mobility as an integral part of CRN, this work has addressed the lack of QoS support for VoIP applications during spectrum handoff in CRN. Accordingly, an integrated spectrum handoff algorithm is suitably designed in conjunction with channel allocation strategies, that not only guarantees successful handoff to the idle channels but also improves the QoS of existing VoIP calls. The detailed design of real-time spectrum handoff algorithm and channel allocation policies are proposed and implemented successfully to ensure handoff along with QoS guarantees for ongoing VoIP calls. Initially, the spectrum handoff algorithm is developed for the SU through the proposed *VAST*, *ProReact* and *Early Call Acceptance* policies. Channel selection and allocation strategies are designed in the next phase by modeling the Target Channel Sequence calculation problem as a fractional knapsack problem and solving it in optimal *O (n log n)* time with the proposed *GA_TCS* algorithm by the Spectrum Controller node. Analytical expressions for important system metrics like idle channel selection probability, channel search time, handoff delay and dropping probability are derived and simulated for analysis. Observations reveal a minimum of 10% reduction in call drop probability with average handoff delay below 150 ms. It also ensures atleast 180 ms of effective time slot for VoIP communication for every CR cycle of 250ms, thus enhancing the QoS of VoIP call.

The proposed design methodology in this study initiates new dimensions in research with respect *to Call Admission Control (CAC)* mechanisms [7.28], weighted *Queuing Models* and *Heterogeneous network scenarios* comprising of RT and NRT users. *Introducing channel failures and their recoveries* is another aspect to be studied in respect to the proposed design. Most importantly, considering the enormous significance of VoIP and CRN in the context of *5G networks*, this work can be extended to facilitate their implementation by addressing the challenges involved in 5G networks.

The innovative research credentials of this study have duly been recognized with the ***Young Scientist Award –First Prize jointly given by IEEE and URSI in the IEEE RADIO Conference held at Mauritius, 2015.*** In addition, the outcome of this study has been published in ***IOP Journal' 16*** (***Thomson Reuters-Web of Science and Elsevier Scopus Indexed***) and also in the ***Conference Proceedings of IEEE RADIO'15***. An extended version of this work has also been communicated to ***IEEE Transactions on Mobile Computing'16 (SCI Indexed)***.



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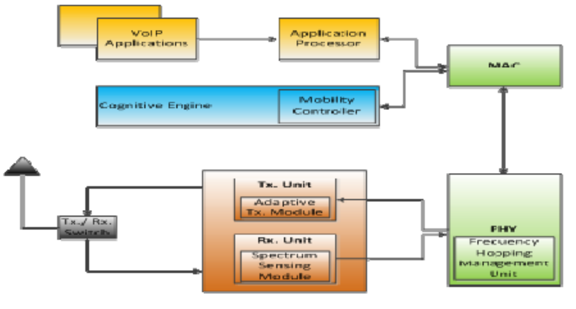
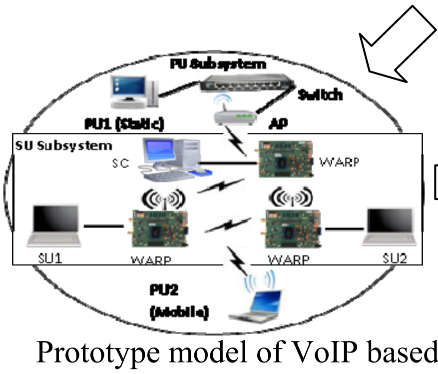
Chapter 8.

PROTOTYPE MODELING OF COGNITIVE RADIO USERS WITH QOS AWARE DESIGN POLICIES FOR VOIP COMMUNICATION

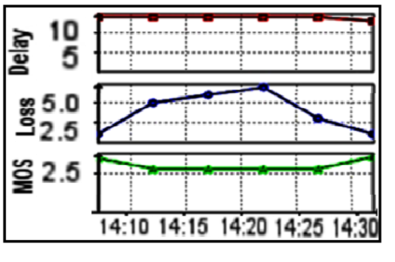
Chapter Highlights

Functional Model for VoIP SU





Prototype model of VoIP based CRN in test-bed



QoS for VoIP Call by SU

CHAPTER 8: Prototype Modeling of Cognitive Radio Users with QoS Aware Design Policies for VoIP Communication

"I think that in the discussion of natural problems we ought to begin not with the Scriptures, but with experiments, and demonstrations."

-Galileo Galilei

Outline of the Chapter

It is evident from the previous chapters that the joint tuning of parameters and policies in both VoIP and CR domain are absolutely crucial towards sustaining long duration VoIP calls in next generation emerging networks like CRN. Analytical and simulation studies in these chapters have also established the credibility and performance superiority of the proposed algorithms with respect to earlier works in literature. However, the feasibility of these formulated algorithms can only be achieved through practical implementation in a hardware test-bed, which will validate these research findings and render them suitable for widespread deployment. In addition, while dealing with VoIP applications, QoE (as discussed in Chapter 2) assumes utmost importance alongside the QoS metrics. As QoE aims to evaluate the end-user satisfaction during VoIP sessions, it can only be measured through real test-bed implementation of VoIP communication over a generic test-bed for

CRN. However, test-bed implementation of VoIP calls in CRN is a highly complicated and challenging aspect that requires thorough investigations of the hardware modules and the related software codes.

Considering the level of complexities involved in such design operations, it is imperative that only few research works as in [8.1, 8.2] have ventured into test-bed modeling aspects concerning real-time applications. However, these works have modeled CR users under some generic assumptions, which limit the practical applicability of such models in real world applications. In this aspect, this chapter focuses on the complete design and characterization of VoIP SU prototype by eliminating the drawbacks in the existing works.

The critical components of modeling a VoIP user include i) adaptive CR policies with minimal complexity; ii) real-time feedback based cross-layer architecture with optimal parameterization; and iii) well-defined interfaces among the functional blocks to ensure loose coupling with the underlying hardware. ***Taking all of them into account, the contribution of this chapter includes the robust design and implementation of CR user test-bed model through optimal parameter configuration and design policy formulation governed by a modified CR principle.*** These policies interact via customized interfaces with the ultimate focus on ensuring target QoS for VoIP SUs without disrupting PU traffic. Experimental operations resulted in a good quality VoIP call with a negligible PU packet loss ratio and rise in call acceptance ratio for the SUs. Thus, both QoS and QoE are satisfactorily maintained through the design methodology in this work.

8.1 Introduction

The practical implementation of real-time VoIP transmission in CRN requires development of a complete real-time communication system that will include both the PUs as well as the SUs. While PUs are required to generate traffic according to different application-centric traffic patterns, the SUs must be equipped with cognitive functionalities along with configured parameters that will be sufficient enough to provide guaranteed QoS to the VoIP applications. This must also be supported by a real-time monitoring tool to measure and evaluate the QoS metrics such as delay, jitter, packet loss and

MOS as well as the network traffic including PU and SU throughput and packet loss due to interference at the PU ends. In general, the work in [8.3] has clearly pointed out the essential aspects of any device-to-device communication including capability negotiation and location management with the objective of saving time and bandwidth resources, and also uses a performance evaluation parameter to monitor the feasibility of such communication. Till date, reports on such development of actual VoIP based CRN communication systems are scarce in the literature and are listed as follows.

8.1.1 Literature Survey

Considering the design complexities involved in QoS aware policy formulation and CR modeling, it is obvious that only few works [8.4-8.6] have indeed studied CR systems in test-bed. To cite them, firstly a Dynamic Spectrum Management Engine is developed in [8.4] to perform Call Admission and channel monitoring functions in CR. On the other hand, the work in [8.5] facilitates dynamic control on the transmission of individual SUs with the inclusion of frequency hopping radio unit that switches carrier among many frequency channels using a pseudo-random sequence. Another work in [8.6] establishes “cognitive” aspect in CR system with incorporation of three classes of cognition models, namely i) Radio Environment Model, ii) Mobility Model, and iii) Application Model.

Specifically, two noted works have been reported in literature, namely the RECOG Model in [8.1] and the Soft real-time model in [8.2] that deal with real-time applications in CR test-beds. These models are analyzed as follows.

RECOG [8.1] is a VoIP based Model for CRN that has been developed in the test-bed. The main focus of this work is to equip the Access Points (APs) with CR functionalities that include sensing, dropping and handoff decisions. Individual SUs that are involved in VoIP communication are not involved in CR methodology apart from transmission, thus SUs are not intelligent CR users by definition. Moreover, RECOG model considers two transceivers for each SU, one for sensing and the other for transmission and acts in infrastructure based mode. CR functionalities are implemented using the Wireless Open Access Research Platform (WARP) [8.7] boards. Although this model successfully maintains R-Factor [8.8] within a range of 30 to 60 and provides a good quality VoIP call, it suffers from several drawbacks. Firstly, considering cost-effectiveness as one of the primary factors for implementing VoIP, having two transceivers for every user in a network is not a commercially feasible solution. Moreover, such a system will also suffer from energy efficiency issues considering sensing by multiple radio units and will thus struggle to maintain the standards of modern "green" communication. Also, allotting the decision-making functionalities with respect to sensing, dropping and handoff to the centralized AP in contrast to individual SUs decreases the cognitive capability of the users in true sense. Thus, this model is not suitable for adhoc mode of communication where all CR users may form the networking. Again, in case of failure of the centralized AP with sensing capabilities, the entire communication will be disrupted. Finally, this model does not take into account the intricacies of VoIP call signaling protocols as well as performance characteristics for PUs. Thus the RECOG model design is more focused to CRN than the integrated system as discussed later in this Chapter.

Another Soft-real-time test-bed based model over CRN has been devised in [8.2]. This model provides real-time support to video streaming applications over CRN. Each individual SU is a CR user whose methodologies are developed using USRP (Universal Software Radio Peripheral) boards in conjunction with GNU Radio [8.9]. A major drawback of this model is that the SUs have first-hand knowledge of the frequency hopping based PU traffic behavior, which is a very specific case of CRN. Therefore, such a model is not a generic one and is not commercially viable for widespread acceptance where PU arrival is unpredictable and needs proper estimation and detection.

In order to overcome the problems associated with these models, a more complete and robust communication system for VoIP applications in CRN is required to be implemented in a generic test-bed and this forms the primary motivation behind the work in this chapter. This outcome of this work will establish the feasibility of real-time VoIP communication in CRN, and will further guarantee the widespread acceptability of the proposed model for application development and research studies.

8.1.2 Significant Contributions

The contribution of this study includes the robust design and implementation of CR user prototype in a real test-bed. Experimental derivation of threshold parameters related to sensing and transmission renders reliability and practical applicability to the designed model whose outcome can henceforth be used as reference in both simulation and test-bed related research activities in this field. The specific contributions of this study are described as follows.

1. An extensive literature survey in Section 8.1 has already highlighted the design complexities involved in test-bed implementation of CR user prototypes. Subsequently, the motivation for this study has also been established.
2. Thereafter, the enhanced design principle for CR users is proposed in Section 8.2 that will be followed by every SU implementing real-time VoIP applications in CRN. This design principle encompasses all the core aspects of CR cycle and modifies them so as to minimize the delays in interaction among the different functional blocks of SUs.
3. The Functional Model for SU is designed following this design principle in Section 8.3. All the essential modules including the Application Processor, Cognitive Engine, Transceiver blocks, etc. are incorporated in the model following a cross-layer architecture.
4. A generic CRN test-bed is subsequently developed in Section 8.4 using different hardware and software components, that incorporates both PU and SU prototype models with adaptive tuning of different parameters (on-off timing intervals, activity rates, transition probabilities from busy to idle states and vice-versa, etc.).
5. The Prototype model for Real-Time SU is equipped with optimal parameters and proposed design policies and further deployed in the test-bed for VoIP communication in Section 8.5. Thereafter, the SU prototype performs long duration VoIP calls under different activity levels of PU traffic, based on which the critical system metrics are monitored and thoroughly analyzed.

6. Comparative performance analysis is performed (both qualitatively and on a quantified basis) in Section 8.6 that establishes the performance superiority of the designed test-bed model in this chapter over previous works in the literature. Novelty of the work in this chapter is subsequently derived.

Finally, the chapter is concluded in Section 8.7.

8.2 Enhanced CR Design Principle

Considering the QoS requirements of any real-time VoIP communication, every SU must take decisions related to spectrum access in minimal time while satisfying two primary objectives; i) maintaining the call quality at an acceptable level (by reducing the interruptions in communication), ii) minimizing interference with PU transmissions, licensed on that particular frequency band. It is hence deduced that the traditional CR system [8.10] comprising of fixed sensing and transmission intervals will fail under these circumstances. By a traditional CR system, we mean the conventional fixed "Sense" before "Talk" policy based CRN which is followed in most of the published works in literature. Consequently, the modified CR principle for enhancing the cognitive capabilities of VoIP SUs is underlined as follows.

In the Initialization phase, the PU traffic is generated based on standard traffic models, along with the selection of VoIP parameters for SUs (codecs, retransmission limits, call signaling protocol, etc.). This is followed by determining the benchmark values of sensing and transmission durations for successful VoIP communication. After a default channel is allotted to the VoIP SU, it enters the Spectrum Sensing phase, where it performs energy detection based spectrum sensing for PU detection. The minimum energy threshold level to detect PU presence is carefully obtained following trial runs, in order to minimize the effects of false alarm and miss-detection. The entire sensing operation is completed within the pre-determined sensing duration, which is adaptively varied based on the network conditions.

On securing an idle channel for transmission, the SU analyzes the channel conditions in the Spectrum Analysis phase and accordingly, adjusts its transmission interval for transmitting the voice packets. The sensing results along with the calculated transmission intervals are incorporated in the Spectrum Decision phase, where a methodology is adopted to decide whether to i) transmit in the current slot in the selected channel, or ii) drop from the existing channel and perform spectrum handoff. In the first case, the SU enters the Transmission phase and performs communication by adaptively varying its transmission durations. In the latter scenario, the SU performs spectrum handoff in the Spectrum Mobility phase to move to another channel, following a pre-determined Target Channel Sequence (TCS).

It is noteworthy that delay should be contained during three phases, namely; i) call setup time in Initialization Phase, ii) call session time originating in the Spectrum Sensing phase and terminating in the Transmission phase, and iii) call resumption time during handoff in Spectrum Mobility phase. Hence, the entire call session is monitored and logged with real-time monitoring tools in the Monitoring phase. The overall design principle is illustrated in Fig. 8.1.

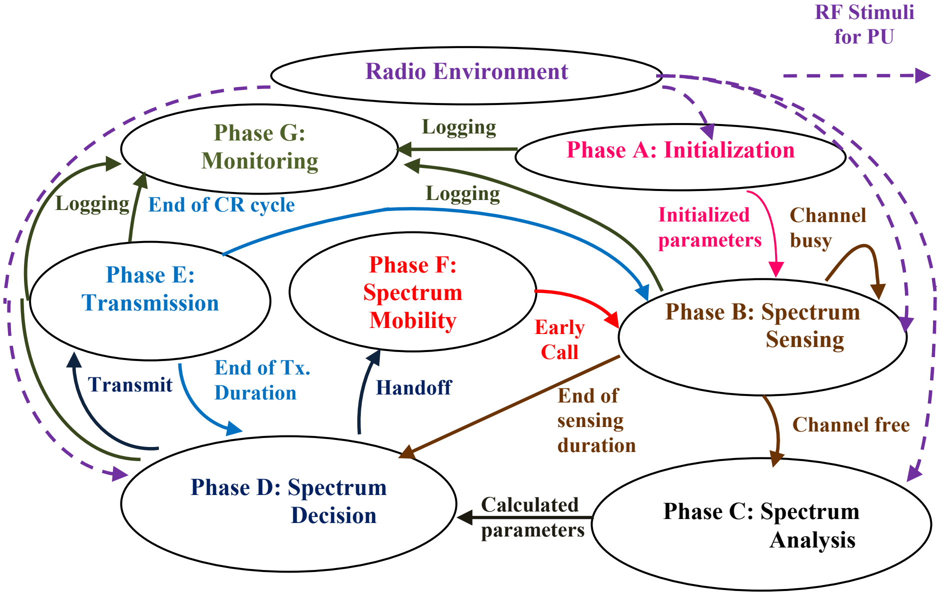


Fig. 8.1 Schematic Diagram illustrating the proposed CR principle for SUs

From the experimental perspective related to practical deployment of VoIP SU in CR test-bed, the input parameters comprise of the PU traffic characteristics and the related network conditions. The output parameters include VoIP QoS metrics (MOS, Packet loss%, throughput, handoff delay, call acceptance ratio, energy efficiency) and PU metrics (Packet loss%, throughput). The methodology includes all the design policies as formulated in Chapter 7 (sensing, transmission, dropping, handoff, logging, etc.) and related parameter configuration. This is illustrated as a schematic diagram in Fig. 8.2. It must be noted that Chapter 7 has already performed mathematical formulation and subsequent simulation studies and established the efficiency of the proposed policies. This study is carried forward in this chapter with test-bed design and implementation.

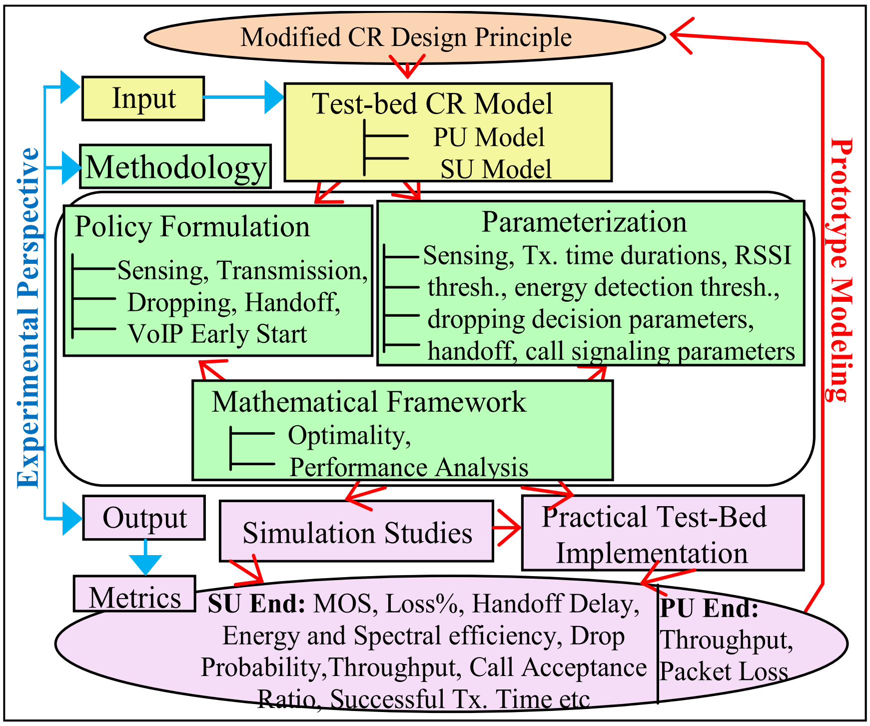


Fig. 8.2 Complete Design Approach for real-time SU Prototype Modeling in CRN

8.3 Prototype Design for Real-time SU

Based on the formulated policies in Chapter 7, the design of a complete SU prototype is modeled in this section using cross-layer design approach that abides by the aforementioned CR principle. The design model of SU prototype comprises of several functional units that coordinate with each other in jointly executing those policies, and is demonstrated in Fig. 8.3.

Firstly, the Application Processor performs all the application management and processing tasks particularly involved in VoIP applications, such as call signaling, call management and monitoring, session management, etc. The Cognitive Engine imparts cognitive functionalities to the SU terminal, including the management and monitoring of ongoing VoIP transmissions and intelligent decision-making features. It also includes the Mobility Controller that performs spectrum handoff to a new channel when the currently occupied channel becomes unavailable due to several reasons such as PU arrival, harmful interference with other users, channel deterioration, etc. The MAC layer of the SU terminal interacts with both the Application Processor and Cognitive Engine and further communicates with the PHY layer for actual transmission.

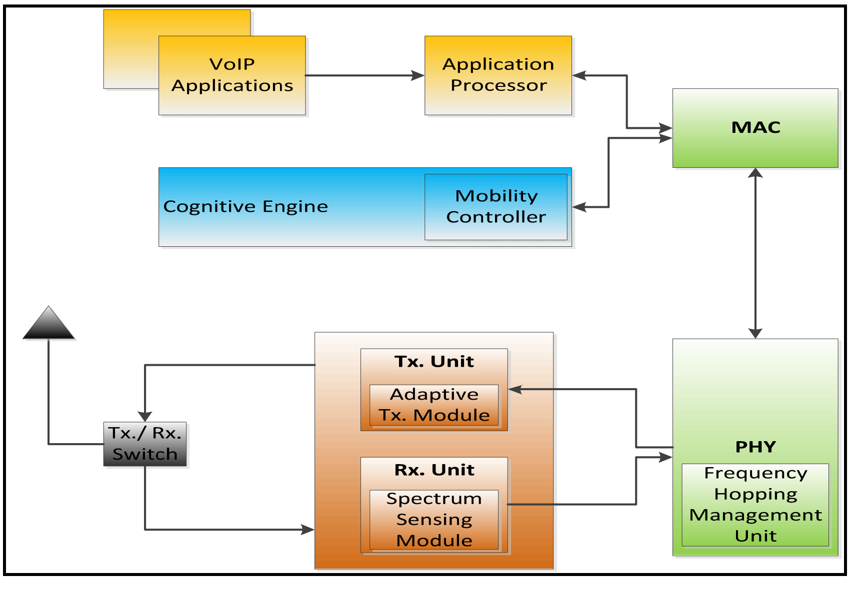


Fig. 8.3 Design Model of SU (CR user) Prototype along with its components

The PHY layer is attached to the transceiver of the SU element. The transceiver apart from performing actual transmission and reception incorporates an Adaptive Transmission Unit in the Transmitter section that adaptively varies the transmission characteristics based on information exchange with the PHY. The Receiver Unit in the Transceiver unit includes the Spectrum Sensing Module that performs the energy detection and senses the availability of a particular channel free from any PU activity for possible transmission. The sensing information is passed to the PHY layer and eventually to the MAC layer for enabling necessary decision-making actions. The PHY layer also incorporates Frequency Hopping Management Unit that enables the transceiver to switch the operating channel based on the developed handoff methodology. Thus, the SU terminal supports cross-layer architecture for better adaptability and efficiency. The flowchart detailing all the functionalities of the SU terminal is shown in Fig. 8.4.

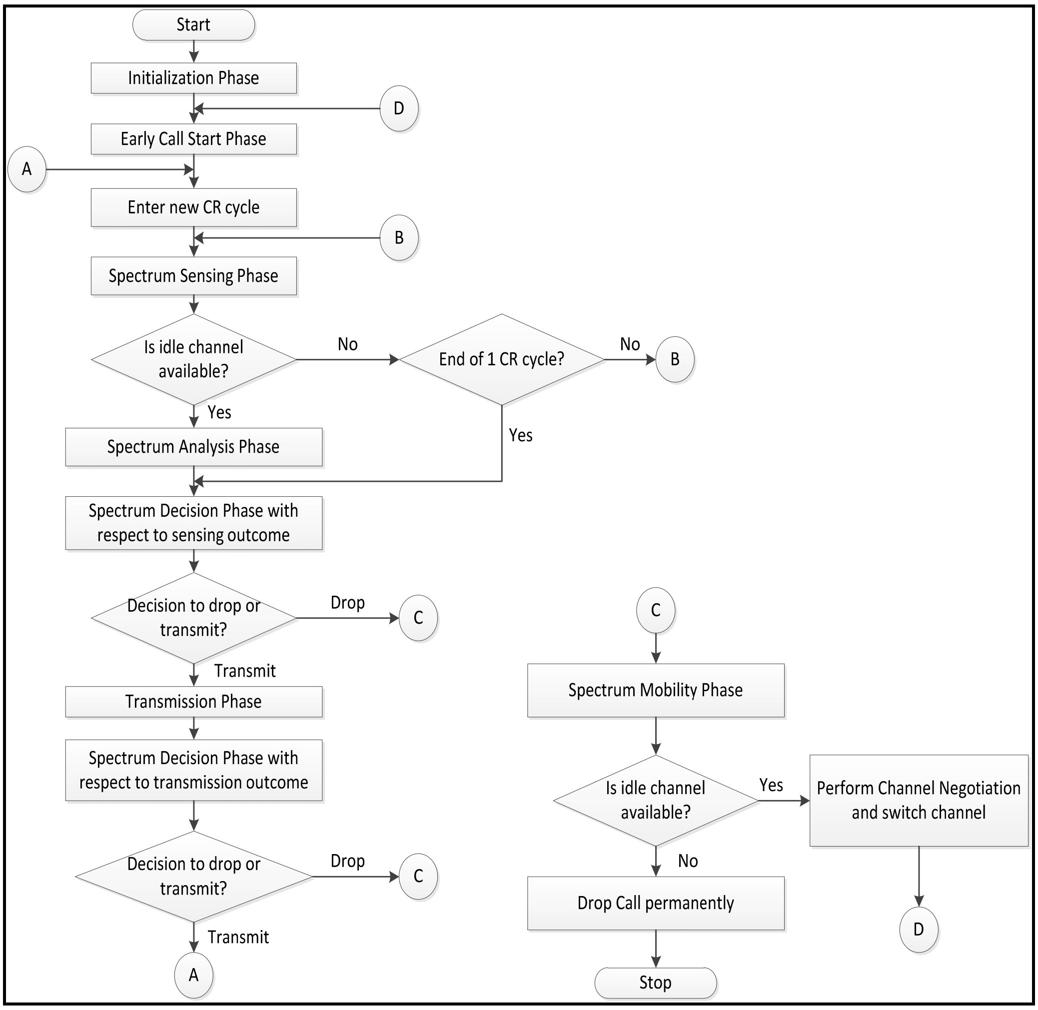


Fig. 8.4 Flowchart depicting all the functionalities as embedded in the SU Prototype Model

8.4 Test-bed Model for CRN

In order to validate the design prototype of CR user, a test-bed based system model of the CR system involving PUs and VoIP SUs is developed and successfully deployed. The test-bed comprises of both hardware components (such as WARP v2 and v3 SDR kits [8.7], Wi-Fi enabled computers, operating Wi-Fi network) and software elements (including SJphone softphones, Wireshark, Network Emulator for Windows Toolkit etc.). A general overview of these components is initially provided in this section, followed by the detailed description of the CR Test-bed Model.

8.4.1 Overview of Test-bed Components

The hardware test-bed model comprises of both VoIP and Cognitive Radio modules that have been established using different hardware and software tools. All these equipments have been suitably configured to ensure the opportunistic mode of communication for the SUs while supporting the stringent QoS requirements for the VoIP applications.

(i) Wireless Open-Access Research Platform (WARP) Board

The CR platform is primarily implemented with Software Defined Radio (SDR) boards where cognitive functionalities are embedded as per the design methodology. The SDR based Wireless Open-Access Research Platform (WARP) [8.7] is used for this purpose.

WARP a scalable and extensible programmable wireless platform, built from the ground up to prototype advanced wireless networks. WARP combines high-performance programmable hardware with an open-source repository of reference designs and support materials. The WARP comprises of the FPGA boards along with clock and radio boards, WARPLAB repositories, OFDM Reference Designs, Medium Access Layer components, etc.

Mango Communications WARP v3 is the latest generation of WARP hardware, integrating a Virtex-6 FPGA, two programmable RF interfaces and a variety of peripherals. The WARP v3 board can support any Virtex-6 FPGA in the FF1156 package. It must be noted that most of the WARP v3 boards are built with Xilinx part XC6VLX240T-2FFG1156C (LX240T device, speed grade -2, lead free 1156-pin package, commercial temperature range). The WARP v3 board is depicted in Fig. 8.5.

The Radio Board is an integral component of WARP kit. It comprises of a RF Transceiver that is connected to the WARP board using daughter card connectors. The radio board has two antenna ports. Both are fitted with SMA jacks (standard polarity female connectors). The two ports are connected to a DPDT RF switch. The other side of the switch is connected to the transmit and receive paths leading back to the RF transceiver. The antenna switch is controlled by a 2-bit digital signal, driven by the FPGA. The architecture is illustrated in Fig. 8.6.

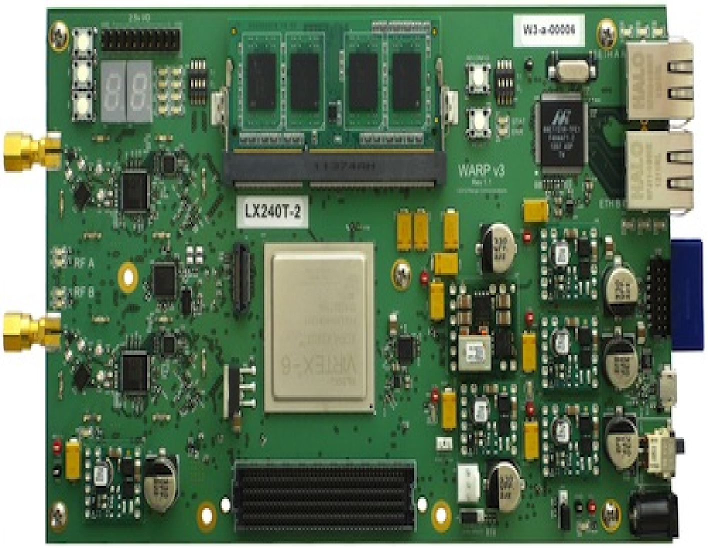


Fig. 8.5 WARP v3 SDR Board with radio daughter cards

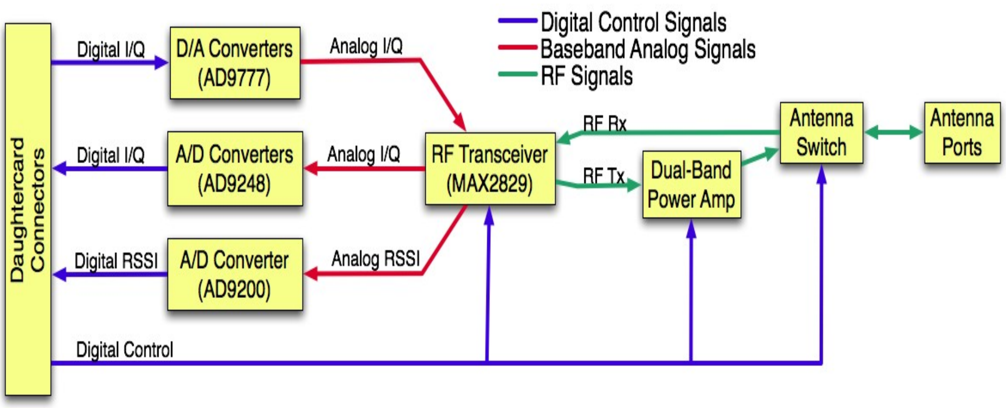


Fig. 8.6 WARP Radio Board

The WARP OFDM Reference Design uses this Radio Board and implements a real-time network stack on a WARP node. The design includes a MIMO OFDM physical layer and flexible MAC interface for building custom protocols. This design demonstrates the full MAC/PHY capabilities of WARP. All processing (hardware control, signal processing, MAC protocol) is executed in real-time by each WARP node. PCs can be attached to WARP nodes via Ethernet, but are used only for traffic generation/analysis.

Each release of the WARP design integrates known-good, interoperable versions of every component. All of the MAC and PHY components are open-source and are available in the repository. The design consists of the following components and is highlighted in Fig. 8.7.

- MAC Application: Top-level C code implementing a wireless MAC protocol.
- WARPMAC framework: Low-level PHY control and MAC primitives, implemented in C code.
- OFDM PHY: FPGA implementation of the OFDM physical layer, built in System Generator.
- Support Peripherals: Other peripheral cores in the FPGA (timer, radio bridges, etc.)

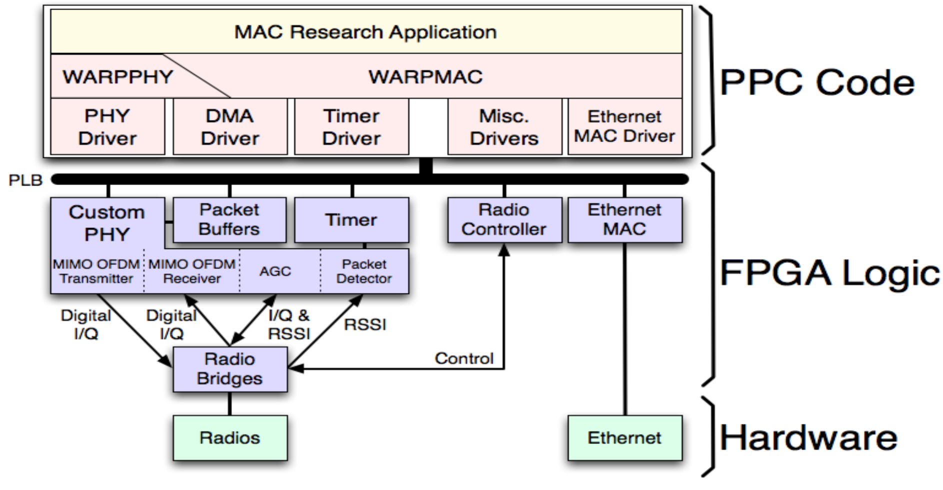


Fig. 8.7 WARP OFDM Reference Design

(ii) ***SJPhone Softphone***

After the SDR based CR platform, the next component of significant importance is the VoIP application launcher, that is, a softphone which can generate and manage VoIP communications. Accordingly, SJPhone is used for this purpose.

SJphone [8.11] is a VoIP softphone that allows users to speak over Internet with any desktops, notebooks, PDAs, stand-alone IP-phones and even with any conventional landline or mobile phones. It supports both SIP and H.323 industry standards, and is fully inter-operable with most major Internet Telephony Service Providers (ITSP) and software and hardware manufacturers.

Different interfaces of SJPhone are depicted in Fig. 8.10.

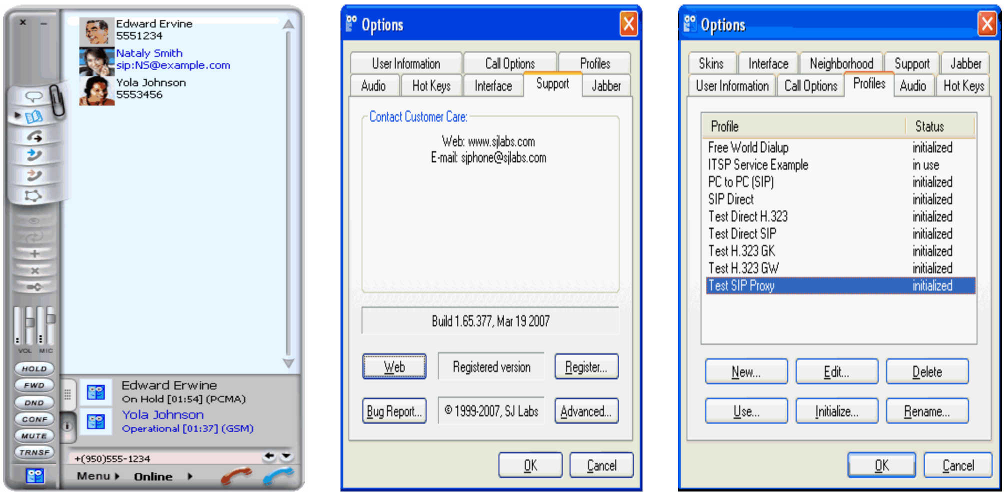


Fig. 8.8 SJPhone Interface

Advanced users may build their own private VoIP networks using an H.323 Gatekeeper, SIP proxy, IP-PBX, and other VoIP components. They can also connect to conventional telephones using H.323/SIP Gateway. SIP features include TCP support, TLS tunneling support, full support for SRV/NAPTR records and transport layer enhancements, and also DNS support in the SIP stack. Prominent H.323 features include customized service profiles that allow users to create their own profiles for calls through H.323 Gatekeeper, Gateway, or SIP Proxy. Some advanced features include the provisions for multisession calls and manual codec selection, support for extended H.323 address syntax, QoS/TOS/DiffServ support, and advanced lost packet recovery solution offering better sound quality over a poor connection.

(iii) ***ManageEngine VQManager***

Proper monitoring and logging of VoIP sessions is crucial towards evaluating the real-life performance of the design methodologies in the CR domain. Accordingly, ManageEngineVQManager is used as the VoIP monitoring tool in this work.

ManageEngineVQManager [8.12] is a powerful, web-based, real-time QoS monitoring tool for VoIP networks. It enables IT administrators to monitor their VoIP network for voice quality, call traffic, bandwidth utilization and keep track of active calls and failed calls. VQManager can monitor any device or user-agent that supports SIP, Skinny, H.323 and RTP/RTCP. Data flow in VQManager is depicted in the figure below.

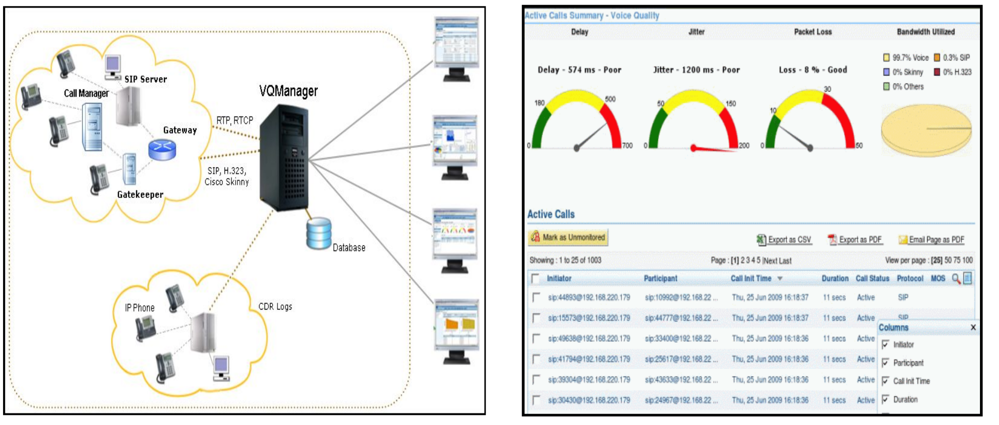


Fig. 8.9 VQ Manager Web Interface

Some significant features of VQManager are discussed as follows.

- Proactive, continuous monitoring of the QoS and bandwidth of VoIP Network.
- Alarm generation based on incomplete calls, answer delay, voice bandwidth usage and other metrics.
- Pictorial representation of call flow plotting all SIP, Skinny and H.323 requests from call start to the end.
- Real-time monitoring and trend analysis for troubleshooting.
- Importing call information using CDR files such as FTP access or uploaded via HTTP.
- Bandwidth usage graph with split up between voice and non-voice data.
- Filtering criteria to capture or monitor call traffic from specific IP addresses.
- Information on 'what is going on' in VoIP network and 'how it performs' are presented in the form of comprehensive, intuitive and informative reports.

(iv) *__Wireshark__*

Although VQManager captures the call details and the related QoS metrics, we need another component in the test-bed that can monitor network related data such as throughput, packet information, timestamps and related parameters. Wireshark is the most appropriate software for this purpose and is hence used in this work.

Wireshark [8.13] is a network protocol analyzer. It lets us capture and interactively browse the traffic running on a computer network. It has a rich and powerful feature set and is world's most popular tool of its kind. It runs on most computing platforms including Windows, OS X, Linux, and UNIX. Network professionals, security experts, developers, and educators around the world use it regularly. It is freely available as open source, and is released under the GNU General Public License version 2. The advantage of using Wireshark is it can read live data from Ethernet, Token-Ring, FDDI, serial (PPP and SLIP) (if the OS on which it's running allows Wireshark to do so), 802.11 wireless LAN, ATM connections, and the "any" device supported on Linux by recent versions of libpcap. A screenshot of the Wireshark interface is depicted in Fig. 8.11.

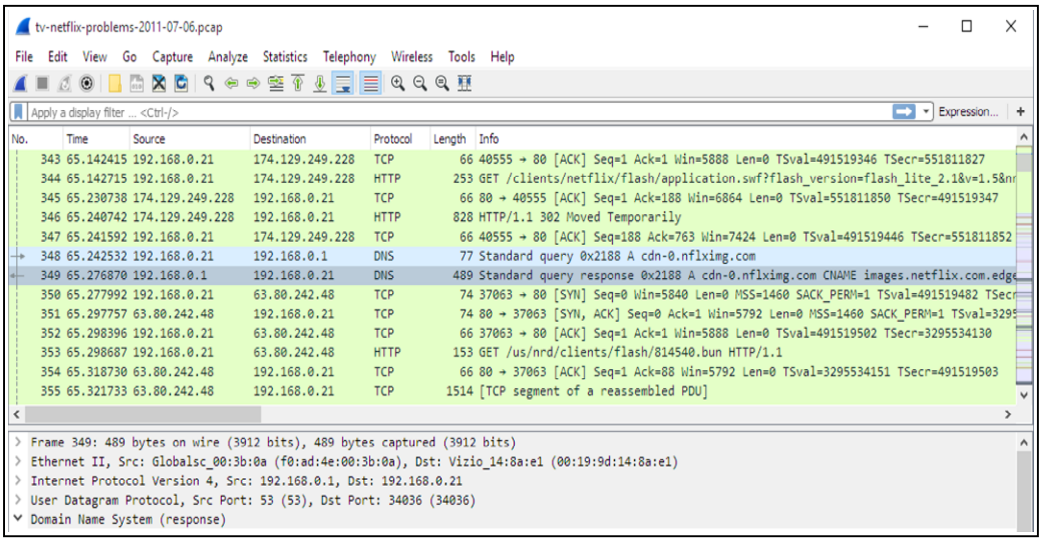


Fig. 8.10 Wireshark Monitoring Interface with detailed network information

Some intended uses of Wireshark are enlisted as follows.

- Network administrators use it to troubleshoot network problems.
- Network security engineers use it to examine security problems.
- Developers use it to debug protocol implementations.
- People use it to learn network protocol internals.

(v) Network Emulator for Windows Toolkit

Finally, another software is used in conjunction with Wireshark, namely the Network Emulator for Windows Toolkit (NEWT) for two reasons.

Firstly, NEWT also measures real-time network data such as throughput, packet loss, etc. and thus, gives us with another set of readings with which we can verify the outcome of Wireshark. Secondly, NEWT is also a powerful emulator that can emulate different types of network conditions through real-time modifications in the generated data, and thus can be applied to test different system conditions.

NEWT [8.14] is a software-based emulator that can emulate the behavior of both wired and wireless networks using a reliable physical link, such as an Ethernet. A variety of network attributes are incorporated into the NEWT emulation model, including round-trip time across the network (latency), the amount of available bandwidth, queuing behavior, a given degree of packet loss, reordering of packets, and error propagations. NEWT also provides flexibility in filtering network packets based on IP addresses or protocols such as TCP, UDP, and ICMP.

NEWT can be used by product testers and network-based application developers to assess performance, predict the impact of change, or make decisions about technology optimization. When compared to hardware test beds, it is a lot cheaper and a more flexible solution in testing network-related software under various network conditions.

The architecture of NEWT is shown in the figure below.

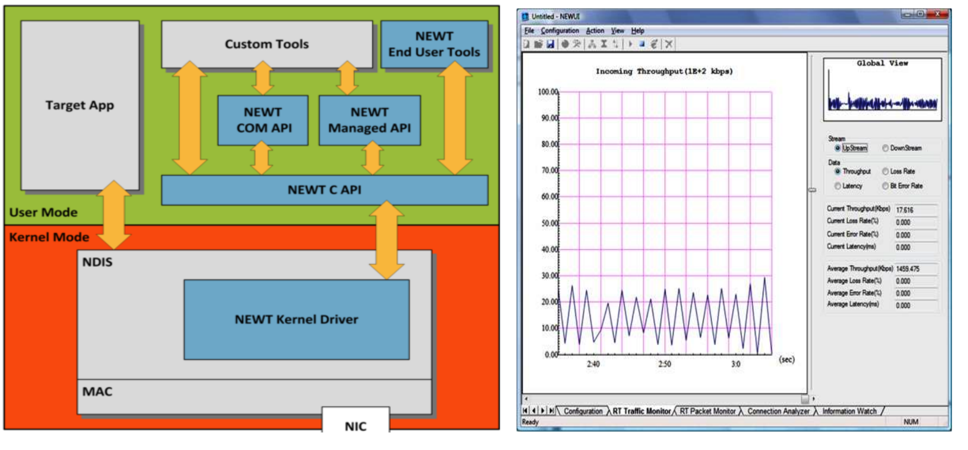


Fig. 8.11 NEWT Architecture with Monitoring Interface

NEWT UI creates a default channel for an emulator. The channel being edited will be highlighted in the channel navigation bar. Using NEWT UI, a user can monitor emulation by three ways, namely, i) viewing runtime traffic statistics, ii) observing packet exchange history, and iii) viewing the connection information.

8.4.2 CRN Model Design in the Test-bed

The CR system is implemented in the test-bed using the aforementioned components and comprises of both SU and PU subsystems as shown in Fig. 8.12. The SU subsystem comprises of one or more SC nodes that manage several SUs under their domain. SDR based WARP board is used to implement the cognitive capabilities of the SUs. As WARP boards integrate programmable hardware (high-performance FPGA with flexible RF peripherals) with open-source repository of reference designs, these enable us to develop appropriate modules for sensing, transmission, dropping and handoff operations in suitable code-blocks, towards building the implementable version of the SU Prototype as described in Section 8.3.

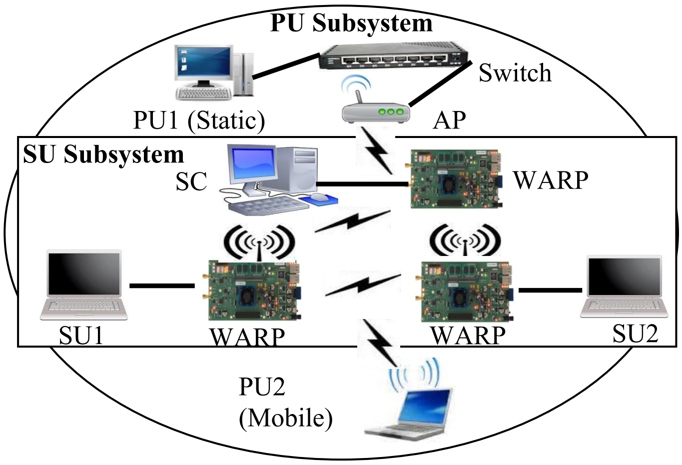


Fig. 8.12 Prototype model of VoIP based CRN in the test-bed

Additionally, WARP is also used to implement the SC with the design of complicated code-blocks that perform target channel selection, channel allocation and communication with the SUs. The SU prototype implements SJPhone softphone in the application layer that performs VoIP communication among the SUs. These softphones are connected to the WARP kits via Ethernet links.

As WARP operates in the unlicensed ISM (2.4GHz) band, an external Wi-Fi network is developed with HP ProCurve Switch and MSM310 Access Points [8.15] to serve as primary network as depicted in Fig. 8.12. PUs comprise of a Wi-Fi enabled laptop (resembling a mobile user) and a standard computer connected via Ethernet to the switch (denoting a static user). The tools used for monitoring and logging purposes include ManageEngine VQManager, WireShark and NEWT, all of which have been described in the previous section. It is worth mentioning that the VoIP calls are recorded for 30-40 minutes on an average to capture the variation in network dynamics with random PU activities.

8.5 Implementation of CR Policies in the Designed Test-Bed Model

The formulated policies in Chapter 5 are now implemented by the SU Prototype in the designed CRN Model and evaluated for performance efficiency with respect to earlier works.

8.5.1 PU Traffic Model

Exponential on-off traffic model [8.16] is considered for the PU, where the transition rates from idle to busy periods and vice-versa are calculated based on the activity factor. For example, an activity factor of 0.2 indicates that a PU remains in the on-state for 20% of the time (denoted by t_{on}) and switches off to the idle state (denoted by t_{off}) for 80% remaining time, where both the on and off periods vary exponentially. The actual values of t_{on} and t_{off} for a chosen activity factor are generated randomly from a uniform interval created around the calculated values for t_{on} and t_{off}. Three different activity factors are selected, that correspond to a network with high, medium and low PU activity respectively.

Additionally, these transition rates are associated with distinct probabilities in order to generate a variable traffic model corresponding to best case, superior case, inferior case and worst case scenarios. Therefore, the mean transition rates for each scenario j are calculated as,

$$\alpha_{mean}^{(j)} = \sum_{i=1}^{3} \alpha_i \Pr^{(j)} \tag{8.1}$$

$$\beta_{mean}^{(j)} = \sum_{i=1}^{3} \beta_i \Pr^{(j)} \tag{8.2}$$

The pdfs for busy and idle states are given by,

$$f_{busy}^{(j)}(t) = \alpha_{mean}^{(j)} e^{-\alpha_{mean}^{(j)} t} \tag{8.3}$$

$$f_{idle}^{(j)}(t) = \beta_{mean}^{(j)} e^{-\beta_{mean}^{(j)} t} \tag{8.4}$$

where α_i, β_i = transition rates from *busy* to *idle* periods and vice-versa for *i*th activity factor, and $Pr^{(j)}$=Probability that *j*th scenario occurs.

A TCP-IP client server model is applied for packet generation and transmission by the PUs, since such a model helps in obtaining the total number of packets lost due to interference (and hence, retransmitted). Experiments are performed to record the mean PU data rate for all the scenarios, and the same is illustrated in Table 8.1.

A screenshot of the PU traffic monitoring tool (WireShark) that measures both the PU throughput and the number of packets lost/retransmitted in an ongoing transmission is depicted in Fig. 8.13.

Table 8.1 Summary of PU Activity Model in the Test-Bed

	Activity Factor	**Transition Rates**		**Time Periods (in ms)**		**Probabilities**			
		α	**β**	**t_{on}**	**t_{off}**	**Best-Case (Scenario I)**	**Superior-Case (Scenario II)**	**Inferior-Case (Scenario III)**	**Worst-Case (Scenario IV)**
Low	0.2	0.5	0.125	180	720	0.7	0.5	0.4	0.3
Medium	0.3	0.33	0.142	270	630	0.2	0.4	0.3	0.3
High	0.5	0.2	0.2	450	450	0.1	0.1	0.3	0.4
Recorded Mean Data Rate for each Scenario				in Mbps		1.48	1.69	2.23	2.59
				in pps		125	140	196	210

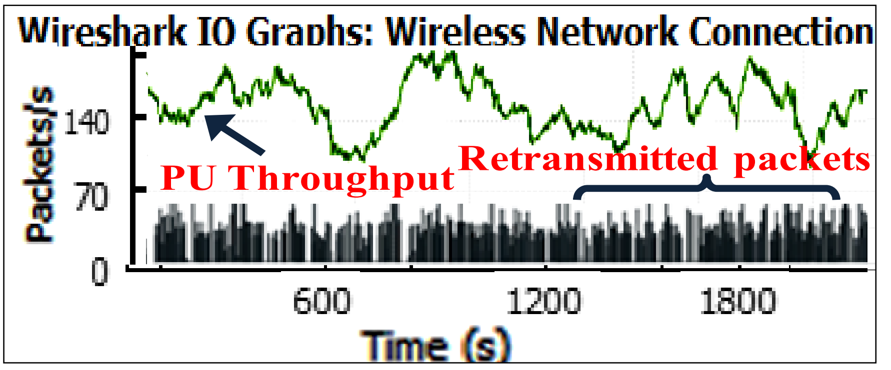


Fig. 8.13 WireShark Screenshot depicting PU throughput and subsequent packet loss due to interference with concurrent SU transmissions

8.5.2 SU Traffic Parameters

The SU establishes VoIP communication via the standard H.323 call-signaling protocol using the SJPhone softphone. Open-source Internet Low Bit Rate Codec (iLBC) [8.17] is used as the voice codec that supports robust communication over IP networks with improved MOS compared to other codecs (such as G.729A). A frame size of 30 ms is considered that yields 13.33 kbps of data rate (33.33 pps). The QoS metrics that are monitored real-time by VQManager in the Monitoring phase include delay, loss and MOS.

8.5.3 Optimal Parameter Configuration

Optimal parameters for SU are selected using the analytical framework of Chapter 7 and is defined as follows. "Select parameters (t_{ls}, t_{hs}, t_{ld} and t_{hd}) optimally such that the actual transmission duration as given by (7.43) in Chapter 7 is maximized subject to the constraint that SU packet loss ratio is <= 0.05". The *three-phase spectrum handoff* algorithm as described in Chapter 7 is implemented in the SU Prototype model. Threshold values are obtained as follows: t_{ls}=0.3 ms, t_{hs}=3 ms, t_{ld}=41 ms, t_{hd}=410 ms. It is experimentally observed from the monitoring tool (Fig. 8.14) that during ongoing VoIP communication by the SUs, packet loss marginally crossed the threshold limit (6%) on increasing the sensing duration to 3 ms while MOS degraded to 2.7.

At the same time, decreasing the transmission duration for each SU beyond 41 ms drastically increased the packet loss to 25% with a minimal MOS value of 1 as depicted in Fig. 8.14. Therefore, the benchmark values for sensing and transmission by SU in a normal VoIP communication were chosen as 3 ms and 41 ms respectively. However, the optimal call quality was achieved with a sensing time of 0.3 ms and transmission duration of 410 ms in Fig. 8.14 where the average value of MOS remained above 3 and packet loss decreased to 2%, thus justifying our initial selection. In addition, t_{s_i} is obtained as 0.15 ms.

Fig. 8.14 Different VQManager Screenshots depicting packet loss for SU under low, optimal and high transmission durations respectively

The RSSI (Received Signal Strength Indicator) value above which PU presence is successfully detected is set to 2500, based on which the threshold power is calculated as -83.15 dBm. The reason for selecting a low RSSI threshold is to ensure that even a weak PU signal is detected. It is found that any possible PU presence is successfully detected when the sensed energy is greater than or equal to 0.00029 pJ, 0.00044pJ and 0.0044pJ during t_{s_i}, t_{ls} and t_{hs} respectively.

Mathematically, the probability of sensing a channel to be busy by SU is given by,

$$\Pr_{\widehat{busy}} = (1-\Pr_{\hat{e}<e|busy})\Pr_{busy} + \Pr_{\hat{e}>e|idle}\Pr_{idle} \qquad (8.5)$$

where $\Pr_{\widehat{busy}}$ denote the busy channel probability as sensed by SU, $\Pr_{busy}$, $\Pr_{idle}$ = actual channel busy and idle probabilities, $\Pr_{\hat{e}<e|busy}$ = Probability of miss-detection (energy of sensed channel is less than threshold, even though

channel is busy), $\Pr_{\hat{e}>e|idle}$ = Probability of false alarm (energy of sensed channel is greater than threshold, even though channel is idle). Thus, it is clearly observed from (8.5) that the selection of RSSI threshold (2500 in this case) is very crucial to eliminate the effects of imperfect spectrum sensing.

8.5.4 Adaptive Transmission as per *VAST* Policy

Every participating SU prototype model is initially equipped with the adaptive sensing and transmission method in *VAST* policy (described in Chapter 7) by incorporating suitable modules in the PHY and MAC layer. It is observed in Table 8.2 that VoIP communication is successful in terms of high throughput and MOS and low packet loss percentage in Scenarios I and II, when PU activity is low. From the PU perspective, negligible packet loss is witnessed in Scenarios I and II. However, as PU traffic throughput increases in Scenarios III and IV, VoIP call quality degrades with respect to MOS and packet loss percentage. Also, increased interference between PU and SU is recorded in terms of high packet loss ratio in the worst-case scenario.

Table 8.2 Test-Bed Output of SU Prototype after implementing Adaptive Transmission (*VAST*) Policy

Scenarios	SU VoIP Call Metrics						PU Metrics	
	Packet Loss %		MOS		Mean Throughput (in kbps)	Comments	Throughput (in pps)	Packet Loss %
	Max.	Avg.	Max.	Avg.				
Best-Case	6	5	4.2	3.8	25.792	Fine	125	0.007 (negligible)
Superior-Case	7	5	3.6	2.9	23.268	Tolerable	137	0.015 (negligible)
Inferior-Case	11	8	3.1	2.8	20.591	Degrading	196	0.016 (negligible)
Worst-Case	14	9	2.9	1.5	19.003	Poor	201	0.01

Overall, these SUs have adaptively varied their sensing and transmission durations and improved MOS value with decrease in the PU activity (Fig. 8.15). Compared to a conventional CR cycle with fixed sense and transmit slots [8.10], the developed method has performed better with improved MOS for SU and low packet loss ratio for PU as seen in Fig. 8.16.

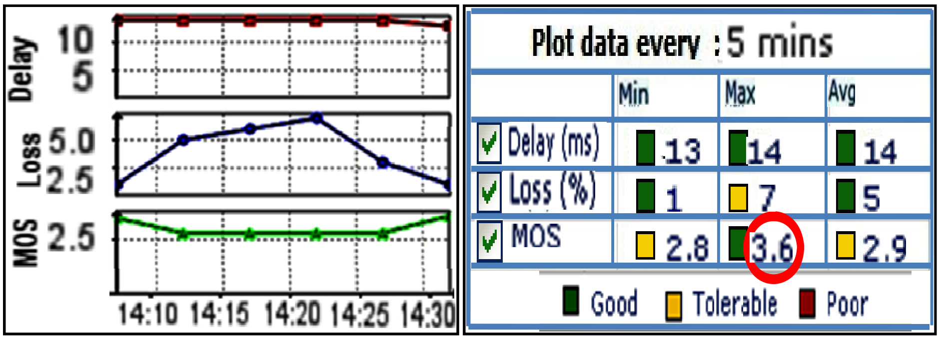


Fig. 8.15 VQManager Screenshots depicting adaptive variation in QoS metrics under adaptive transmission (*VAST*) policy

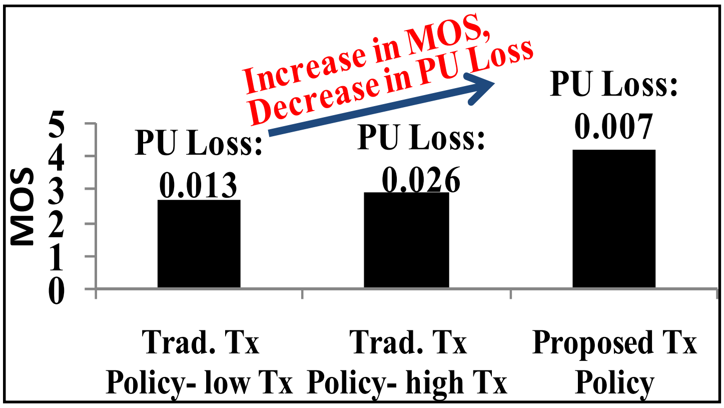


Fig. 8.16 Performance Superiority of the Proposed Design in this work over traditional CR Studies in literature

With respect to energy consumption by SU, measurements in the WARP board pointed to power utilization of 16.8 W and 16.2 W for the transmission and reception modes respectively. Under the adaptive sensing and transmission method, it is found that MOS and spectral efficiency (ξ) have increased in the t_{hd} interval compared to t_{ld} duration (ξt_{hd} =1.28 kbps/MHz, ξt_{ld} =1.14 kbps/MHz). However, this has reduced the energy efficiency (η) as SU has shifted from low to high transmission duration (ηt_{hd} =0.01 kbps/mJ, ηt_{ld} =0.076 kbps/mJ).

8.5.5 *Three-level Dropping Decision* Operation

It is inferred from Table 8.2 that the transmission policy in *VAST* fails during the worst-case scenarios with high PU activity, resulting in call quality degradation for SUs and increased packet loss ratio for PUs. As per the overall objective of sustaining VoIP call without hampering PU activity, the SU performs *three-level dropping decision* policy (as explained in Chapter 7) to drop from the current channel.

The outcome of the decision mechanism is recorded via the "putty" console (serial monitor) of the WARP kit during active VoIP calls under adaptive PU traffic. It is observed from Fig. 8.17 that in the worst case scenario, the dropping decision is significantly determined by *Cons_sense*, which decreases as the scenario improves. As PU traffic intensity is lowered, *Alt-tx* gains prominence compared to the other decisions.

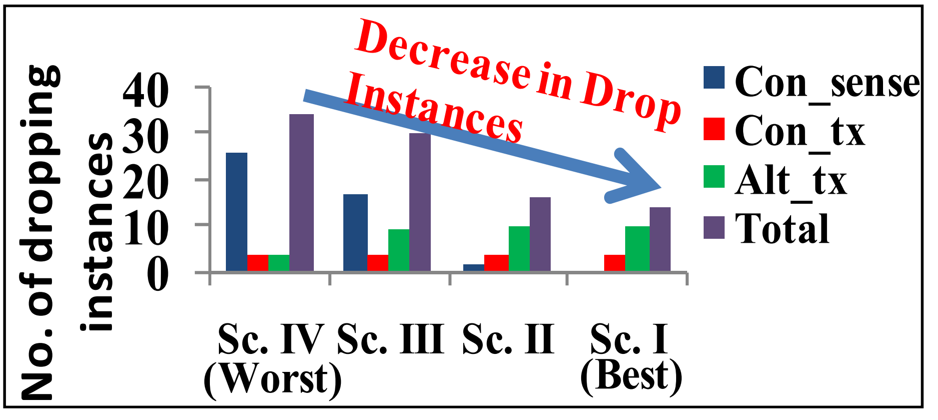


Fig. 8.17 Performance of the proposed *3-level dropping decision policy* in the implemented test-bed model under different scenarios of PU Activity

Probability of occurrence of *Cons_tx* is low as it is overshadowed by *Cons_sense* for high PU activity and *Alt_tx* during low PU traffic. Overall, the number of dropping instances is reduced as one proceeds from the worst to the best-case scenarios.

8.5.6 *Two-Phase Spectrum Handoff* Operation

It has been observed from Fig. 8.18 that SUs have invoked the developed decision method and decided to drop from the current channel under Scenario IV. Subsequently, they perform timely handoff based on the Initiator-Follower based *ProReact* spectrum handoff policy (described in Chapter 7) and re-establish communication on the new channel successfully. The total handoff time (including the channel switching, sensing and consensus time) is further affected by the delays due to the device impairments in WARP kit. The mean value of this delay is recorded to be 1.8s in WireShark and the handoff process is illustrated in Fig. 8.18.

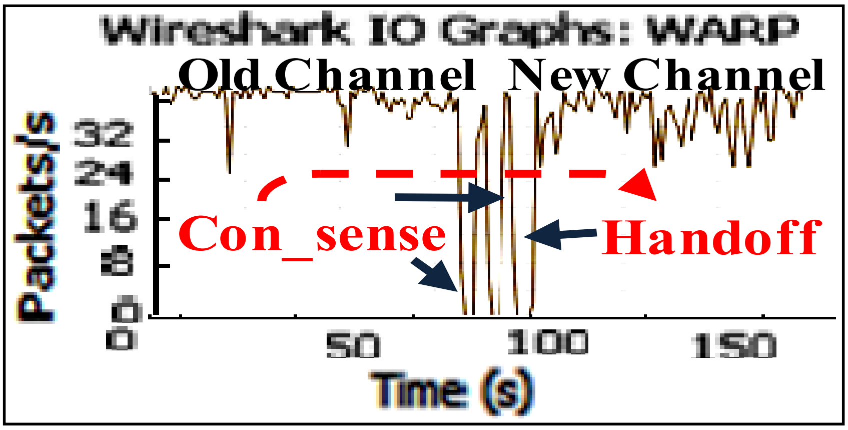


Fig. 8.18 Dropping decision induced by *Con_sense* and subsequent proactive handoff, along with SU throughput as monitored in Wireshark

At the same time, VQManager readings have detected a minimum MOS of 2.7 (2.8 avg.) and maximum packet loss of 10% (5% avg.) during handoff along with a minimum degradation during handoff mechanism (Fig. 8.19). The PU packet loss is also reduced to a negligible 0.005%. Thus, the call quality has improved significantly along with reduced interference to PU traffic after performing the handoff (compared to the MOS value of 1.5 and PU packet loss ratio of 0.01 without handoff as recorded in Table 8.2- Scenario IV).

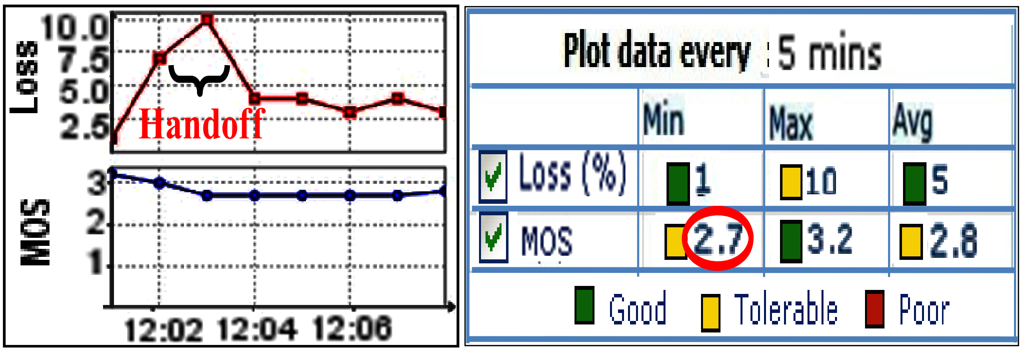


Fig. 8.19 VoIP MOS and Loss% in VQManager under the Spectrum Handoff Condition

8.5.7 Early Call Resumption Operation

Finally, the Early Call Acceptance method as designed in Chapter 7 is incorporated in SU for successful resumption of VoIP call in the new channel after spectrum handoff. The outcome of the policy in Table 8.3 witnesses a marked increase in call acceptance ratio with respect to 20 calls recorded under the Superior and Inferior Case scenarios, as compared to default CR policy with fixed cycle parameters [8.10] and under H.323 Fast Start mechanisms [8.18] in CR environment.

Table 8.3 Performance Improvement in Call Acceptance Ratio in the Designed Model over existing Call Initiation Policies

Scenarios	Default call [8.10]	H.323 Fast Start [8.18]		VoIP Early Call Policy	
	Acceptance Ratio	Acceptance Ratio	Improvement	Acceptance Ratio	Improvement
II	11/20	11/20	0%	15/20	36%
III	4/20	6/20	50%	10/20	150%

8.5.8 *GA_TCS* based Target Channel Allocation by SC

Based on the PU activity model as described in Section 8.5.1, ten different channels are designed with varying busy/idle characteristics as highlighted in Table 8.4. The corresponding throughput of the PUs in these channels is measured using Wireshark and NEWT. Accordingly, the SC calculates the TCS for the first SU based on the designed *GA_TCS* algorithm as discussed in Chapter 7. It is clearly evident in Table 8.4 that the obtained test-bed result closely resembles the ideal TCS outcome.

Table 8.4 Modeling Of Channel Parameters based on Table 8.2 and subsequent TCS selection in SC Node

Channels	1	2	3	4	5	6	7	8	9	10
Probability Set	(4,3,3)	(5,4,1)	(7,2,1)	(8,1,1)	(5,3,2)	(3,3,4)	(2,3,5)	(4,4,2)	(6,3,1)	(1,2,7)
Measured PU Throughput (kbps)	2290	1769	1582	1520	1940	2609	3010	2054	1633	3642
Calculated Channel Sorting List based on increasing throughput: {4,3,9,2,5,8,1,6,7,10}; SC detected Channel Sorted List based on GA_TCS: {3,4,9,2,5,8,1,6,7,10}; TCS for first SU: {3,4,9,2}; TCS Values: Total Busy Time = 709 ms(SC), 740 ms(Actual); Total Idle Time = 3287 ms(SC), 2982 ms(Actual)										

However, minor variations are observed between the actual outcome and ideal outcome primarily due to two reasons: i) hardware constraints in WARP limit the efficiencies of the transceivers during periodic sensing of PU activity; and ii) *GA_TCS* algorithm uses purely exponential on-off traffic model for PUs [8.16], whereas the exponentially derived PU values in the actual test-bed are varied with different probabilities based on a realistic approach.

Therefore, it can be inferred from this section that the configured prototype not only establishes the feasibility of VoIP communication over CRN but also satisfies all the objectives of this work with substantial increase in call quality (MOS-Fig. 8.16), reduction in call drop (Table 8.3), successful spectrum handoff (Fig. 8.19) and minimum interference with the PUs (negligible PU packet loss-Table 8.2). This also implies that the QoE for the VoIP calls (as described in Chapter 2) is successfully maintained for the end-users.

8.6 Comparative Performance Evaluation and Novelty

Comparative performance evaluation has already established superiority of the proposed design policies over the conventional CR model with respect to call quality MOS (Fig. 8.16) and call acceptance ratio (Table 8.3). In addition, the outcome of the conducted experiments in this chapter is also compared with that of the previous works on CR test-beds involving real-time traffic. The novelty of our work is subsequently derived.

Table 8.5 highlights both the similarities of these studies with our work and further highlights the significant drawbacks in those models, which are effectively addressed in our test-bed implementation.

In contrast to the earlier works, three important priority aspects namely 1) Simplicity, 2) Energy Efficiency, and 3) Cost-effectiveness have been imposed in the established model with respect to every SU to ensure widespread acceptability of the developed system. All the methods in the various modules of this invention are developed considering minimal time and low algorithmic complexity that not only reduces the processing power and latency but also ensures simplicity and compatibility across different device categories in use within the system. Additionally, this model utilizes only a single transceiver to perform sensing and transmission using adaptive methodologies, thereby contributing to a significant reduction in energy consumption. This also makes this invention a cost-effective solution for building VoIP supported CR system.

Table 8.5 Comparative Performance Evaluation with existing research studies in literature

Testbed Based Models	Similarities with the proposed model	Key Drawbacks eliminated in our model	Novelty of the work in this chapter
1.RECOG[8.1]	1. Both models focus on improving QoS of VoIP calls in CR environment. 2. Both models manage to keep R-Factor within tolerable limits (30-60) during VoIP transmission. 3. These models also achieve minimum perceivable degradation during spectrum handoff. 4. Both these models deploy feedback based policy to learn from the past channel conditions before making dropping decisions. 5. Both these works implement CR with the help of WARP SDR kits.	1. Sensing is performed by AP and not by the client SU 2. Two transceivers for simultaneous channel sensing and data transmission, as opposed to a single transceiver for a SU in this work 3. Decision to switch channel is initiated by AP instead of being taken by the SU Thus, SUs in [8.1] are not truly cognitive in nature and lack energy efficiency and cost-effectiveness.	1. RECOG achieves 10ms sensing time with respect to AP as the benchmark value beyond which the call quality degrades. On the other hand, this study accurately establishes a maximum 6.3 ms sensing duration limit for sensing operations by individual SUs. 2. The mean sensing duration for SU in the proposed model (0.3ms-3 ms) is more suitable for adaptive transmission policy compared to a fixed 1 ms duration in [8.2]. 3. As sensing and all other policies are implemented by each SU, this work develops a truly cognitive SU and focuses on the aspects of parameter configuration to minimize energy consumption, unlike other works. 4. Unlike RECOG, this model highlights the importance of the underlying call signaling protocol in achieving higher call
2. Soft Real-time CR Test-bed [8.2]	1. Both these models implement CR policies on the	1. USRP in conjunction with GNU Radio is used	

	individual SUs, making them CR aware. 2. Both these works design policies that provide real-time QoS to delay sensitive applications.	to setup CR environment, as opposed to WARP Kit (which is more robust) in this work. 2. Real-time requirements are studied for video transmissions and not for VoIP calls. 3. Frequency Hopping PU behavior is considered to be known to SUs unlike the present model in this chapter where SUs assume no prior knowledge of PU activity. Thus, SUs are not generic as they assume to have first-hand knowledge of PU traffic behavior. This specific CRN is not suitable for widespread acceptability.	acceptance ratio. 5. The current model also monitors the PU metrics (throughput, packet loss) and aims to make the entire SU communication transparent to PU. 6. Finally, CRN test-bed is a generic one in this work with independent traffic models for PUs and SUs without any assumptions. Thus, it establishes the practical significance and validity of the test-bed outcomes.

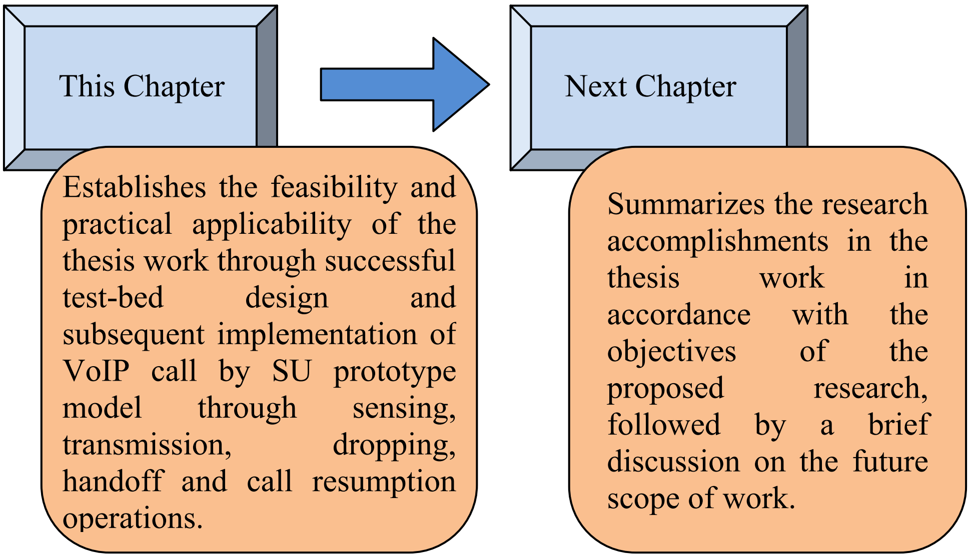
Consequently, the novelty of this work can be ascertained from the fact that equal importance has been meted out to both PUs (in terms of ensuring minimal interference with the SUs unlicensed on that channel) and SUs (by providing guaranteed QoS to their VoIP applications) and thus, the interests of all the users in the developed system are preserved. Therefore, the present CRN model is a generic system that can be applied for providing VoIP as a service using the Cognitive Radio Platform.

8.7 Summary

This chapter provides an innovative insight into the emerging research aspects on application-oriented study of VoIP over CRN, and addresses the critical problem of configuring real-time SUs for VoIP communication with extensive validations in a test-bed model. This paves the way towards determining optimal parameters in CR and VoIP domain in conjunction with suitable design policies based on a modified CR principle to enhance the QoS for CR users without interfering with the PU traffic. A robust CR prototype for the configured SUs is modeled with integrated functional blocks using cross-layer design architecture, and implemented in a generic CRN test-bed for practical significance. Performance analysis of the analytically derived system metrics in the test-bed model confirms 55% improvement in MOS and 73% drop in PU packet loss ratio with 150% rise in call acceptance ratio for the proposed *VAST* and *Early Call Acceptance* Policies. The efficiency of the *three-level dropping* and two-phase *ProReact* policies in the modeled prototype is also justified with the successful test-bed execution of sustained VoIP communication with good call quality (MOS = 3), minimum handoff degradation (maximum 10% packet loss only during handoff) and negligible PU packet loss ratio (=0.005). The novelty of the designed model is also established with respect to conventional CR models and existing research works in this domain.

The developed SU terminal with the incorporation of Application Processor, Cognitive Engine, Mobility Controller, Adaptive Transmission Unit and Spectrum Sensing Module can be implemented in the modern cellular phones after suitably configuring their MAC and PHY layers. This will enable these phones to act as SUs and perform VoIP based communication in the idle frequency bands of other service providers, when their underlying cellular network becomes congested. The service providers too can use this prototype to provide their unused spectrum to the CR users on a temporal basis and at a nominal price. Thus, this model will prove beneficial to the end-users as well as the service providers.

The outcome of this study has been filed for Indian Patent and published in the Indian Patent Journal'16.



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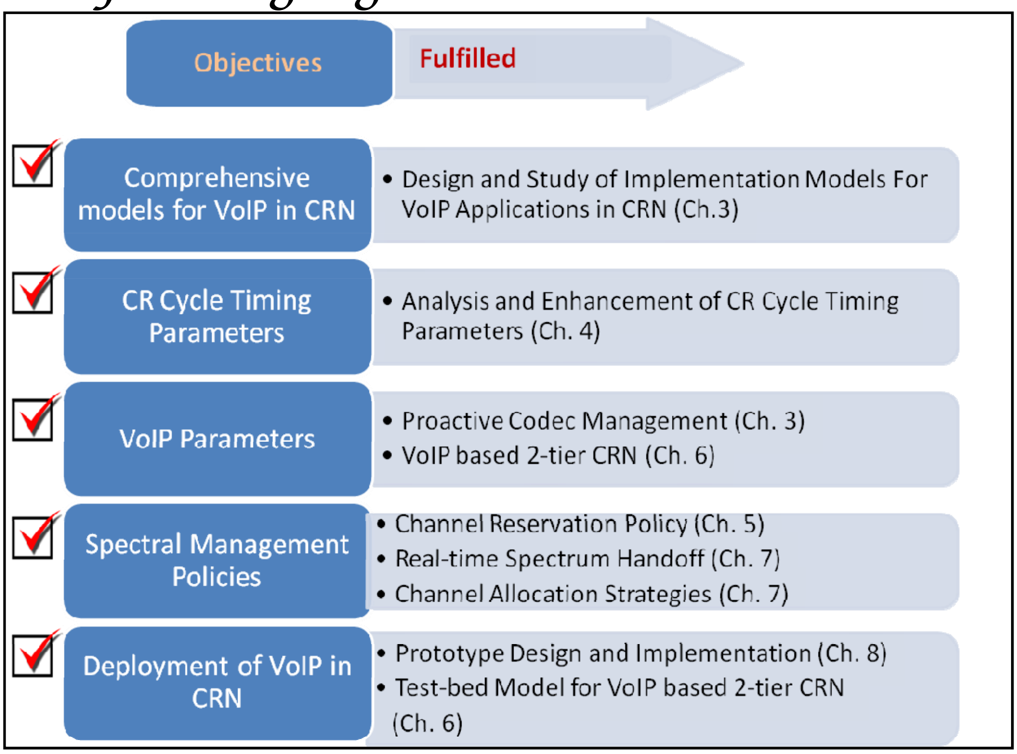
Chapter 9.

CONCLUDING REMARKS

Chapter Highlights



CHAPTER 9: Concluding Remarks

"Quality is not an act, it is a habit."
-Aristotle

Outline of the Chapter

VoIP is indeed the technology of the future that has enthralled one and all in the telecom industry by its rapid development over the years. It has revolutionized the entire telecommunication stream by enabling communication through various multimedia applications in this world of infrastructure convergence. As communication patterns continue to evolve with the introduction of different applications such as MMS, push-to-talk, instant messaging, conferencing, social networking, gaming, and other such services, VoIP offers them at a price that can stand firm even in times of economic recession. While significant enhancements have resulted in successful deployment of VoIP in office and home networks, focus is on increasing user satisfaction (QoE) in terms of reliability and call quality. In this work, the issues relating to performance improvement of VoIP have been addressed specifically under the purview of another next-generation network, namely the Cognitive Radio Networks (CRN).

Evolution in the field of wireless communication has witnessed consistently increasing number of users and wider bandwidth requirement of data and multimedia transmitting technologies that have constantly reduced the availability of frequency spectrum. CRN addresses this problem of spectral congestion by introducing opportunistic usage of the frequency bands that are not heavily occupied by licensed users. In this network, spectrum sensing is done to locate unused spectrum segments and optimally use these segments without harmful interference to the licensed user. Implementation of this technology, therefore, faces unique challenges starting from the capabilities of cognitive radio techniques and the communication protocols that need to be developed for efficient communication to novel spectrum management functionalities such as spectrum sensing, spectrum analysis, spectrum decision as well as spectrum mobility. Despite these challenges, the incentives for selecting CRN as a suitable platform for the VoIP applications can be traced to three factors that are illustrated in Fig. 9.1.

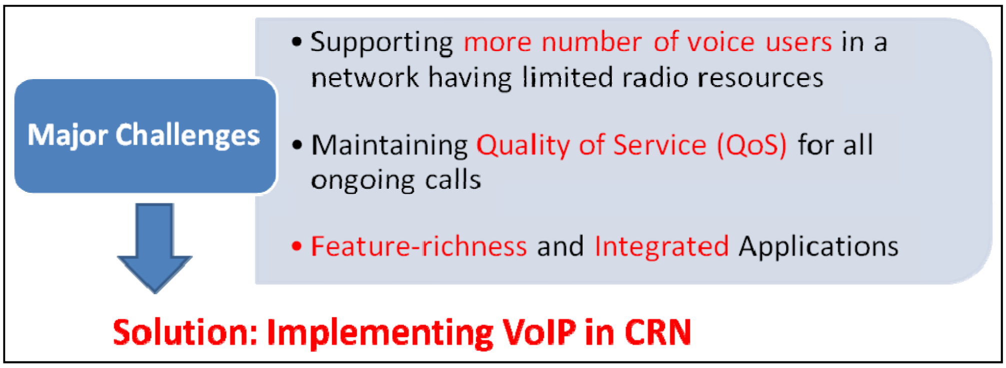


Fig. 9.1Reasons for hosting VoIP applications over CRN

Although considerable progress has been made in the research domain of CRN and its related issues, deployment of real-time applications over CRN has received relatively lesser attention. To be specific, a comprehensive study of VoIP QoS parameters over CRN has not been made till date. Neither do we have a specific CRN management policy for implementing VoIP efficiently over it. This triggers the necessity for active research in this domain in order to exploit the advantages of CRN for the betterment of VoIP and other related real-time transmissions. This also requires modifications in the VoIP domain for its adaptation to the current CRN scenario which varies from other networks in a broad way. This work addresses these issues through robust studies of VoIP communication patterns over CRN both from the systems perspective and in the application domain.

Though the thesis has dealt with enhanced QoS-based spectrum management policies for VoIP services in CRN, the research methodologies in this thesis can be made equally applicable to other real-time applications such as video streaming, multi-party conferencing, online gaming, etc. with suitable modifications as per application requirements.

This chapter provides a brief summary of the research accomplishments carried out in this thesis, in accordance with the overall objectives of research (as stated in Chapter 1), and also discusses the future aspects of system and application development that can be pursued based on the obtained outcome in the thesis.

9.1 Summary of Research Accomplishments

Increasing popularity of VoIP systems has witnessed applications in emerging technologies like CRN. The stringent QoS requirements in VoIP coupled with complexities in CRN have initiated intensive research in the field of performance analysis and optimizations guided by simulation results. However, in the absence of any standard model of VoIP over CRN, accuracy and credibility of simulation output are strongly dependent on proper design of the simulation model that must have a strong mathematical foundation. ***The work in this thesis builds standard models for VoIP in CRN and successfully implements VoIP applications over CRN domain, which serve as the initial point for development with respect to all future simulation studies in VoIP over CRN category.*** Performance study of the models reflects efficient simulation design with sufficient credibility followed by validation of the models mathematically and through comparison of corresponding simulation output. ***Based on these models, a proactive codec adaptation algorithm is proposed that adaptively varies the codec bit rate using active queue management and succeeds in providing the maximum throughput to the SU without interference from the PU.*** Thus, these models can be used by learners and researchers to gather basic knowledge of VoIP transmission in CRN followed by suitable optimizations that are enabled by enormous scope in further development of the designed models.

Deploying VoIP in opportunistic communication models like CRN requires thorough analysis of the factors involved in design and implementation. Consequently, it is observed from simulation studies that the basic CR timing cycle is not worthy of providing adequate QoS for VoIP calls and results in increased PU interference and spectrum handoff delays. ***In order to mitigate this degradation, a two-phase algorithm is designed and implemented in the***

thesis, which introduces momentary sensing slots and adaptively varies the timing durations. Both the end-to-end delay and packet loss for the VoIP sessions are reduced after implementation of the proposed modifications. ***This is followed by the optimal configuration of the transmission time using a novel algorithm before initiating VoIP call in a particular channel.*** This algorithm achieves lower collision rates along with drastic reduction in channel switching instances (and hence lesser spectrum handoff delays). In order to measure the "cognitive capability" of these VoIP users, a derived QoS parameter, as denoted by *cog_cap* is developed in this thesis that uses the call quality metric (R-Factor) of VoIP calls and the occupancy percentage of the wireless channel in CRN for calculation.

A significant research contribution has also been made in the thesis with respect to channel reservation policy for PUs, that reduces the probability of their interference with SUs, though at the cost of lower system throughput. Initially, the system model is defined along with architectural framework to address the practical constraints involved in implementing channel reservation strategy. A novel design approach is then adopted to develop mathematical models, followed by evaluation of network parameters to obtain tradeoff between decrease in interference and low spectrum utilization when channels are reserved in CRN. The optimal number of channels to be reserved is also determined. In addition, as a potential application of channel reservation policy, an analytical framework is designed for VoIP applications in CRN and analyzed in real-life like simulation models which validate the inferences drawn from the mathematical models, recording over 50% improvement in interference-free transmission for SUs with channel reservation scheme in CRN. It is inferred that PU based channel reservation provides adequate QoS to VoIP calls having MOS value more than 3. However, this comes at the price of lower system throughput. ***With the aim to increase this system capacity from the network perspective, this work is further extended by introducing a novel Priority based Adaptive Channel Reservation (PACR) algorithm that comprises of two parts namely, Adaptive Channel Reservation (ACR) where channels are reserved adaptively based on PU activity, and Priority Based Allocation (PBA) which accommodates both real-time VoIP***

SUs and non real-time data SUs in the system based on priority settings. Stochastic analysis in the mathematical models reflects performance improvement in CRN with increased system capacity in terms of SU Sum Goodput. System heterogeneity is further increased as *PACR* algorithm enables CRN to successfully host both VoIP and data applications. Simulation results corresponding to the developed model in OPNET Modeler 16.0.A. confirm more than 100% improvement in throughput for *PACR* based CRN compared to static channel reservation policy.

However, system capacity is bounded by a maximum limit as derived by earlier works in literature. This severely restricts the overall spectral efficiency of the system. ***This problem is effectively addressed by designing a "VoIP based 2-tier CRN" that allows more number of SUs in the system.*** While the first tier of SUs involve in VoIP communication, the second tier of SUs exploits the silence periods in VoIP transmission to send data. In the first phase, analytical models are designed to highlight significant improvement in overall system capacity in 2-tier CRN. Simulation output in OPNET Modeler 16.0.A confirms that 2-tier CRN increases the average link utilization by almost 40% as compared to the Basic CRN for equal transmission rates of both SUtier1 and SUtier2. Markov Models have been designed in the second phase, that have recorded significant reduction in SU dropping and blocking probabilities in spectrum handoff enabled 2-tier CRN along with increase in successful transmission probabilities for SUs. In the third phase, two algorithms based on message passing policy, namely *Simple_msg* and *Periodic_msg* are proposed for practical implementation of VoIP based 2-tier CRN with the aim of increasing system capacity while reducing interference among users. Simulation studies confirm that *Simple_msg* strategy provides higher transmission time for SUtier2 at the cost of increased time of interference and is, therefore, suitable in scenarios with low SUtier1 activity. *Periodic_msg* algorithm, on the other hand, incorporates periodic checking of channel status to reduce interference and is implemented when SUtier1 activity is high. The final plan of work in this regard is to practically implement the novel design concept of 2-tier CRN in the developed hardware test-bed, which completes the design process of the 2-tier based CRN. Accordingly, the principle of 2-tier CRN is applied to the disaster management system and suitably implemented in a generic CRN test-bed. The test-bed model successfully executes VoIP and data applications, and captures a rise of over 450% in system throughput along with 31.13% decrease in energy consumption and 26.23 % increase in spectrum efficiency for 2-tier CRN.

A substantial research contribution has also been made to deal with spectrum handoff instances in CRN. Stringent QoS requirements such as delay, loss and MOS impose unique challenges in CRN when VoIP SU performs this handoff after getting disrupted by sudden PU arrival. Sustaining a VoIP call under such scenario demands joint tuning of sensing, transmission, channel drop decision and handoff functions guided by TCS calculation and allocation policies. ***With focus on this innovative design aspect, this work develops an integrated real-time spectrum handoff algorithm that is executed by the SUs in three parts.*** The first part implements the proposed *VAST* (VoIP based Adaptive Sensing and Transmission) Policy. This part initially performs dynamic configuration of timing intervals by selecting two timing intervals for sensing and transmission durations and adaptively varies them to maximize transmission time, with minimum interference to PU traffic. Based on the configured parameters, the *VAST* policy thereafter implements the *three-level dropping decision* policy. Here it decides to drop from the current channel as per three rules, namely, *Cons_Sense*, *Cons_tx* and *Alt_Tx*. The second part in the handoff algorithm performs QoS aware *two-phase spectrum handoff* policy (denoted by *ProReact*). SU performs this real-time spectrum handoff in two phases. In the first phase, SU executes *Proactive* handoff to an idle channel that belongs to the TCS. However, if every channel in T.C.S. is busy, SU executes the second phase and performs *Reactive* handoff by randomly selecting a suitable channel for sensing and transmission (if that channel is idle). Finally in the third and final part of the handoff algorithm, the SU quickly resumes the interrupted call on successful handoff to prevent call drop due to timeout in H.245 and RTCP. ***While spectrum handoff is performed by the SUs, the TCS calculation is concurrently performed by the centrally located Spectrum Controller (SC) nodes.*** The SC formulates the TCS as a fractional knapsack problem and solves it using the proposed *GA_TCS* Algorithm. Subsequent channel allocation to each SU is performed using three user and channel allocation vectors via a dynamically updated common control channel. While *GA_TCS* selects the maximum number of target channels with the highest cumulative idle time, the three-part spectrum handoff algorithm provides maximum transmission duration of 180 ms (out of the total time of 250 ms) for VoIP calls in average case. It also adaptively ensures that the handoff delay remains below 150 ms during the average-case proactive phase and 600 ms in the worst-case reactive phase.

It is further observed from the literature survey that the existing research studies have performed analytical and simulation based studies to support VoIP communication in CRN. However, the practical significance of all the existing works in VoIP based CRN can only be realized with SU prototype modeling and validation using test-bed implementation. In addition, strict QoS and QoE requirements for VoIP demand adaptive design policies with respect to PU detection, transmission, dropping and spectrum handoff operations followed by their implementation in practical scenarios. ***Considering such complexities in QoS aware policy design and CR modeling, another novel contribution of this thesis includes the robust design and implementation model of a CR system in a real test-bed to characterize the SU for supporting VoIP call through prototype modeling.*** This paves the way towards configuring both CR and VoIP parameters with suitable design policies (spectrum sensing, transmission, decision, handoff, and early start) based on a modified CR principle to enhance the QoS for CR users without interfering with the PU transmissions. The most significant contribution is to determine several threshold parameters related to sensing, transmission, and handoff by an actual test-bed model. Accordingly, the optimal parameters with respect to sensing and transmission are selected using the analytical framework based on which the configured design policies are implemented in the SU prototype. It is observed that the *VAST* policy provides higher MOS for SUs and reduced packet loss for PUs and, thus, performs better than fixed-duration CR timing cycles. Thereafter, the execution of the *ProReact* policy results in a minimum MOS of 2.7, maximum 10% SU packet loss and drastic reduction in PU packet loss ratio to 0.005 while ensuring minimum handoff degradation. Finally, the Early Call Acceptance policy ensures significant reduction in call drops for SUs (over 150%) as compared to existing works in literature. Thus, the configured prototype of SU satisfies all the objectives of this work with substantial increase in call quality (MOS), reduction in call drop, successful spectrum handoff and minimum PU interference (reduced packet loss ratio for PUs).

In a nutshell, the work in this thesis satisfactorily fulfills all the stated objectives through innovative research contributions in the related domains of i) design and test-bed implementation, ii) spectrum management policy formulation with optimal parameter configuration, and iii) extensive studies over analytical frameworks and simulation platforms. This is suitably demonstrated in the following figure.

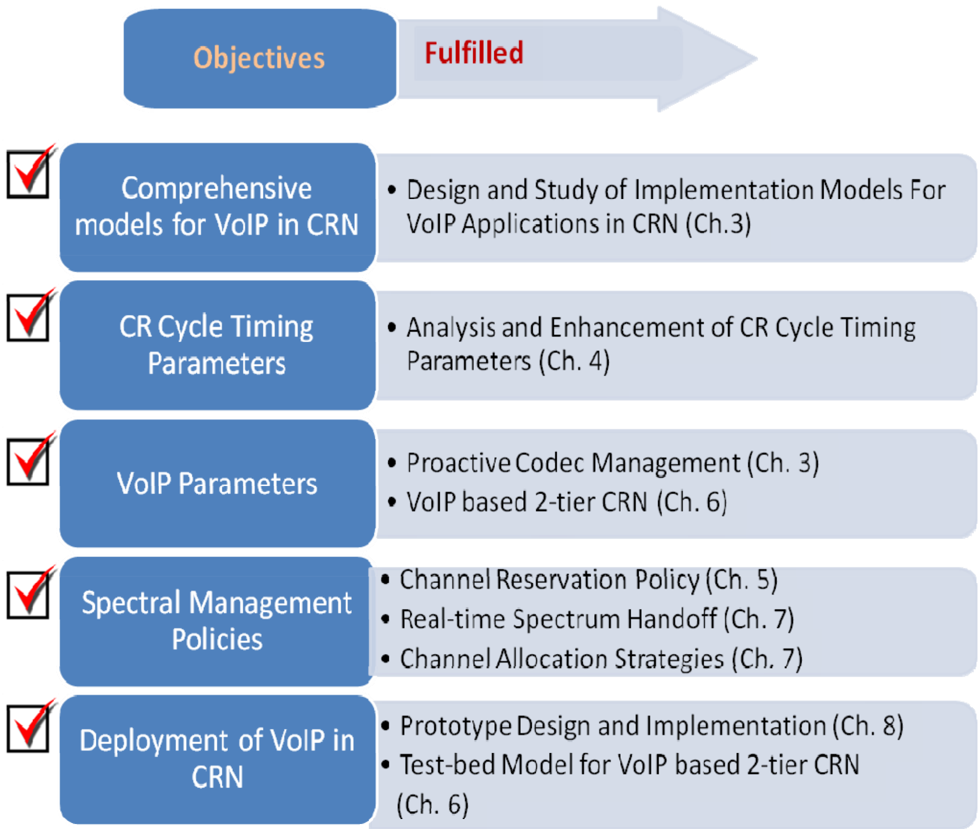


Fig. 9.2 Fulfillment of all the Research Objectives in the Thesis

9.2 Future Scope

This thesis presented the research work to implement real-time VoIP communication over CRN through QoS aware design policies and parameter configuration. In addition, formulating the *Call Admission Control strategy* is essential in such systems which must deploy *different queuing models* to support *heterogeneous applications* with special emphasis on VoIP *and video-streaming services*. In this regard, the research accomplishments in this thesis require further investigation in the following areas as listed below.

1. ***Design of Call Admission Control policy that will determine how many SUs with real-time VoIP/video-streaming applications should be admitted in the system, taking into account several factors such as PU traffic activity, channel conditions, SU priority schemes, energy versus spectral efficiency trade-off, etc.***

2. ***Studies of Heterogeneous CRN where both RT and NRT SUs are admitted and jointly coordinate with each other towards ensuring proper QoS for the RT users, and maximum throughput for the NRT users without causing harmful interference to the PU traffic.***

3. ***Analysis and formulation of a Queuing Model that will be deployed to handle these RT and NRT SUs based on priority settings (for real-time SUs) and preemption policies and suitably configured for situations when two real-time SUs arrive at a same channel at the same time instance.***

4. ***Implementation of video-streaming and conferencing applications over the designed simulation and prototype models that must reconfigure their parameters and tune the existing policies to support both VoIP and video services for the same set of SUs.***

These issues are briefly discussed as follows.

1. ***Design of Call Admission Control policy that will determine how many SUs with real-time VoIP/video-streaming applications should be admitted in the system, taking into account several factors such as PU traffic activity, channel conditions, SU priority schemes, energy versus spectral efficiency trade-off, etc.***

Call Admission Control (CAC) policy is integral to the proper functioning of any networking system and therefore, requires careful attention during the design phase. In CRN, this policy is further constrained by the tradeoff between providing services to the SUs and increasing the spectral efficiency (and system capacity) of the network. When real-time users such as VoIP or video based applications enter the CRN domain and require a default channel for transmission, CAC must determine i) whether to admit them or block them; and ii) on admission, how to assign the default channels to them. Thus channel assignment also becomes a critical aspect to be jointly considered with the CAC mechanism. The efficiency of the CAC policy can be evaluated from the system dropping and blocking probabilities. An efficient design will strive to maintain a balance between these probabilities and is to be studied further as an extension to the research conducted in this thesis work.

2. ***Studies of Heterogeneous CRN where both RT and NRT SUs are admitted and jointly coordinate with each other towards ensuring proper QoS for the RT users, and maximum throughput for the NRT users without causing harmful interference to the PU traffic.***

As CRN is envisioned to host an array of applications by different SUs, it is imperative that deploying a heterogeneous system with respect to applications must be the ultimate objective. One way to generalize the classification criteria of these applications is their real-time requirements. Accordingly, the services that demand stringent QoS in real-time such as VoIP, video-streaming, multi-party conferencing, etc. can be termed as the Real-time (RT) users. Likewise, the applications where data integrity and throughput are the primary concerns can be grouped as Non Real-time (NRT) users. The system metrics in these networks must incorporate the features of both these types of users. For example, the Sum Goodput as extensively used in this thesis must now incorporate these data rates of NRT users with that of the RT users. This thesis has considered such systems during the formulation of *PACR* algorithm (in Chapter 5) and the design of 2-tier CRN (in Chapter 6). In addition, further studies are needed with respect to these SUs in the domain of MAC protocols, channel allocation strategies, energy efficiency policies, etc.

3. Analysis and formulation of a Queuing Model that will be deployed to handle these RT and NRT SUs based on priority settings (for Real-Time SUs) and preemption policies and suitably configured for situations when two real-time SUs arrive at a same channel at the same time instance.

The CRN like any other networking system must also incorporate a robust queuing model for handling these different types of traffic, each having distinct requirements. Two aspects namely, priority and preemption become critical in such model design. Priority refers to the conditions where multiple packets arrive at the queue from different data sources of SUs. Clearly, SUs with RT traffic will have precedence over the NRT ones. But the challenge occurs when multiple RT SUs are involved. The second aspect is the preemption policy where the real-time users must preempt the NRT ones from the queue during selection of the channel. It must be understood that unless these two aspects act efficiently, the system tradeoff between spectral utilization and QoS support can never be obtained. To cite an example, when TCS selects target channels for RT users (as per the proposed methodology in Chapter 7), the target channels at the far end of the queue remain unutilized and hence in this case, QoS prevails over system capacity. On the other hand, if the TCS selection is kept to a minimum to allocate these channels to the NRT users, overall system capacity will increase at the cost of higher call drop counts (due to limited number of target channels available for consideration during spectrum handoff for the RT SUs) and thus the QoS support will be hampered. In this way, queuing models play a major role in heterogeneous CRN and should be studied further in accordance with the designed policies in this thesis work.

4. ***Implementation of video-streaming and conferencing applications over the designed simulation and prototype models that must reconfigure their parameters and tune the existing policies to support both VoIP and video services for the same set of SUs.***

While this thesis has exclusively dealt with VoIP applications in CRN, other real-time applications such as video-streaming applications, multi-party conferencing services, etc. can be studied as extension of the proposed research in this thesis work. Every such application has its own set of constraints in terms of the level of QoS required, the application-level parameters and their interaction with the system. For example, a video codec has different properties compared to the voice codec in terms of the MOS offered, the traffic rate generated and the compression characteristics. Obviously, these parameters require re-configuration. Also, the spectral management and mobility aspects as designed in this thesis require further investigations with respect to these applications. Accordingly, the SU prototype model must be modified to support these different application requirements. The ultimate focus should be to effectively integrate these services with VoIP and deploy them in the designed test-bed with minimal changes to the overall setup.

A generic system model that incorporates all these extended aspects of research is envisioned in Fig. 9.3. This model can henceforth be used for future reference.

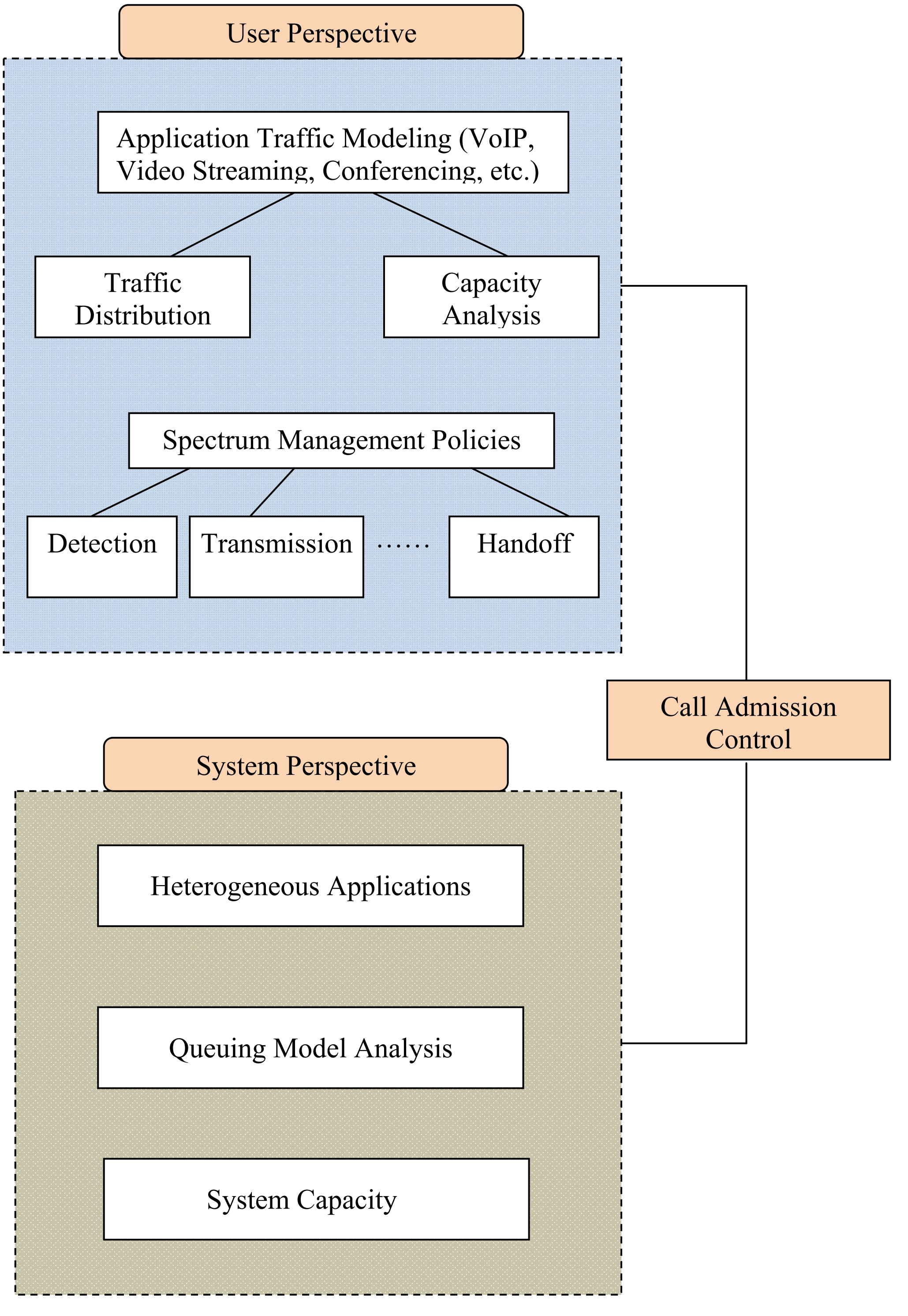


Fig. 9.3 Schematic Model highlighting the future scope of work

It is finally concluded from this chapter that this thesis has succeeded in addressing all the primary objectives of research concerning the feasibility and applicability of hosting VoIP communications over CR Networks. At the same time, the research challenges and their solutions in this thesis have initiated further studies with respect to capacity planning, call admission control, queuing models and support for video and other multimedia applications. The significance of this study will only grow in the coming years as VoIP is slated to become one of the primary modes of communication over the next-generation networks.