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.