Spectrum Access in Cognitive Radio Networks Based on Prediction and Estimation

by

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Abstract

In the literature, Cognitive radio (CR) as well as full-duplex (FD) communication technologies are proposed to increase the spectrum efficiency. The main contribution of this thesis is to introduce prediction and estimation techniques with low control overhead, and use the predicted or estimated information in resource allocation in CR networks, both in the overlay networks and the underlay networks. Prediction and estimation are important in increasing the data rate and keeping the interference at a low level.

In the overlay scheme, I modeled the primary user (PU) traffic characteristics of the channels using the Probabilistic Suffix Tree (PST) algorithm. Then using this PST algorithm, I introduced a frequency hopping based control channel and derived its theoretical properties. Then I proposed two methods for selecting a channel set for transmission, which took into account both the PU channel usage statistics and, secondary user (SU) channel usage statistics as perceived by an SU of interest. The first scheme selected channels having the highest probability of successful transmission, while the second calculated a net reward using a marked Markov chain. Then using simulations, I showed that our scheme caused acceptable interference to the PUs and has better throughput performance, compared to a scheme selecting channels randomly.

Then I proposed two joint channel assignment and power allocation schemes for a bidirectional FD underlay CR network with network assistance. The first scheme used the information on the number of total SU pairs present in the network. In the second scheme, I used least squares based estimation and Kalman filtering to estimate the interference at the monitoring stations using the local interference. It reduced the control overhead of

keeping track of active SUs. In both of these schemes each SU pair decided on the channels to be used in the half-duplex mode and the full-duplex mode using local information. This joint optimization was done running channel assignment and power allocation algorithms alternatively. In the power allocation problem, I used a technique called monotonic optimization. After simulating both of these schemes I showed that the scheme based on estimation performs satisfactorily given that it has less control overhead.

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Chapter 1

Introduction

Wireless communication in the 21^{st} century is growing at an unprecedented rate. One of the major limitations to this growth is the scarcity of the electromagnetic spectrum which is suitable for terrestrial communication. The main authorities in the USA and Canada who do the assignment of frequencies in this spectrum are the Federal Communications Commission (FCC) and the Spectrum and Radio Policy directorate, respectively. These entities assign the spectrum to organizations on fixed long term contracts to operate on a particular geographic region. Although this method worked in the past, vast growth in the wireless services has pushed the FCC to find more efficient spectrum allocation mechanisms to avoid spectrum scarcity [1]. The measurement operation carried out by the FCC's Enforcement Bureau in 2002, for a limited time period gave some insight into the real cause of the widely accepted notion of spectrum scarcity [2]. According to the report of the Spectrum Efficiency working group, although some bands are heavily used there exist bands which have availability in either time and/or frequency and/or space. The recommendations of the Spectrum efficiency working group consisted of promoting flexible use of spectrum, development and deployment of advanced technologies and promoting secondary markets for spectrum [2]. A novel technology called by the names Cognitive Radio (CR), neXt Generation (XG) or Software Defined Radio has the capability to implement those recommendations [1].

Another avenue of increasing spectrum efficiency is by using full duplex (FD) communication [3]. In general communication protocols, the uplink and the downlink communication are done on channels which are orthogonal either in time, frequency or code. In FD communication one device does both transmission and reception using the same frequency band concurrently. Advanced self interference cancellation (SIC) techniques as well as smaller transmission distances in modern networks have enabled devices to use FD communications. FD communication can be used in both overlay and underlay setup (These terms are explained in Section 1.2). However in this thesis we are concentrating on the underlay setup since the full duplex cognitive radios only do continuous sensing while transmitting concurrently.

In what follows in this section we present the concepts, functionalities and the applications of CR. Then in Section 1.7 we present SIC techniques and FD network scenarios present in the literature.

1.1 Overview of Cognitive Radio

As mentioned above, the static allocation of frequency bands gave rise to the problem of "artificial spectrum scarcity" [4]. The term artificial is applied because although the bands are owned by organizations there exist time periods when they are not used. These occurrences are called spectrum holes. The definition of a spectrum hole can be given as: "*A spectrum hole is a band of frequencies assigned to a primary user, but, at a particular time and specific geographic location, the band is not being utilized by that user*" [5]. The term 'primary users' in this definition refers to the entities that have entered into a contract with the spectrum management institutions to use a particular band of frequency in a particular geographic location. These spectrum holes can be exploited with the use of CRs which opportunistically use the spectrum. These cognitive radios are also called by the name Secondary Users (SUs) or Unlicensed users. In order to exploit the spectrum holes, CRs should be able to identify the spectrum bands which are idle at a particular time epoch, select the best available channel, share the channels with other secondary users and to release the channel when the primary user reappears [6]. This was the original motivation of using CRs. This kind of opportunistic using of the spectrum were given the name overlay communication later.

Being an opportunistic user of the spectrum, cognitive radios should have extensive capabilities to operate on diverse frequency bands (eg: GSM900, GSM1800, UHF, VHF), diverse wireless standards (eg: GSM, WLAN, WiMAX) and diverse modulation schemes, and should be capable of switching from one standard to another in a very short time duration [7]. As mentioned above spectrum sensing is one of the main capabilities possessed by an CR. Since the licensed users will be operating on diverse protocols the spectrum sensing is non-trivial. The propagation of signals through the atmosphere causes the signal to deteriorate due to propagation loss, fading and shadowing which makes the problem even worse.

The main characteristics which differentiates a CR from traditional communication devices are cognitive capability and reconfigurability. The capability to detect the availability of usable channels, intelligent power allocation, analyzing the characteristics of the channels and deciding on what bands to use depending on the statistical availability, data rate, bandwidth and transmission mode are called the cognitive capabilities [6]. The reconfigurability refers to the ability of the cognitive radio to change the frequency of operation, the modulation scheme used, the transmission power used and the communication standard used [6]. For example, the frequency of operation may have to be reconfigured due to the reappearance of a primary user on a previously discovered spectrum hole or when a more suitable channel is discovered. This change in frequency band of use is called Spectrum Hand-off. Transmission power control requires the CRs to regulate power below the regulatory maximum to keep the interference to PUs and other SUs low, but at the same time the CR receiver should get a tolerable error rate at the data rate required. Since the power is controlled another SU who is sufficiently far away can use the same channel, thereby increasing the spectrum efficiency. Because of the dynamic nature of spectrum access a

CR should be able to change from one communication standard to another almost instantly when required.

Another important sub-feature in CRs which helps to provide Quality of Service is the "*spectrum mobility*" which combines both the above main features [6]. This feature enables the cognitive radio to reconfigure itself seamlessly so that in a spectrum hand off the applications running on the cognitive radio suffers minimum performance degradation. As one can see, this feature should work on all the layers on the network stack to provide a seamless spectrum hand off. Spectrum hand off means the change of operating frequency band due to the appearance of a primary user, change of spectrum holes due to movement in space or can be due to fading and shadowing. Hence, knowledge of spectrum status and the channel gains between SUs and PUs is critical in this technology.

In the next section we will discuss the two main methods of reusing the PU spectrum in the SU network.

1.2 Overlay and Underlay Networks

In overlay networks a CR senses channels for PU activity and finds out a channel which is not used by a PU and transmits using that channel. If multiple channels are available it uses the best available channel or it can combine the channels using an OFDMA protocol. The term best in the wireless jargon can mean any or all of the features which include: highest signal to noise ratio, highest bandwidth or highest expected idle time left in the channel. The main feature in overlay networks is avoiding interference to the PUs by not using the channels concurrently with them. On the other hand underlay networks uses low power transmissions such that the transmission appears as noise below the transmission threshold to the PU. One of the techniques of achieving low power transmissions is using techniques like Code division multiple access [8, 9] or Ultra wide band radio [10]. Although sounds simple, this should be done with utmost care since increasing the noise floor of the PU above a particular level will immerse the PU signal in noise and make it undetectable to the PU receiver. Another way of achieving the same goal is using power control on the transmissions. Here, techniques like orthogonal frequency division multiple access are used with power control on the sub-carriers. These techniques need information on the channel gains from the devices in the secondary network to the devices in the primary network [11].

In [12], the authors investigate the outage probability at the PU receiver caused by interference in overlay, underlay and interference avoidance underlay SU networks. The interference avoidance underlay scheme also spreads its power over the entire band but it notches out or nullifies the frequencies in which the CR senses PU activity [12]. In this analysis, the authors of [12] found out that, interference avoidance schemes introduce the least interference to the PUs and in the case of imperfect sensing, interference avoidance underlay scheme performs better than the overlay scheme in terms of the interference caused on the PU.

We will discuss the most desirable feature in CR, spectrum sensing, in the next section.

1.3 Overview of Spectrum Hole Detection

Channel sensing is one of the core functions of a CR. In CR, channel sensing involves the collection of PU channel usage information of a channel over a number of dimensions: frequency, time, space and code [13]. Channel sensing in the frequency dimension implies the detection of frequency bands which are not used. In time dimension it implies the detection of temporal variations of a particular band. The space dimension includes the detection of availability of a frequency band at an area given by some distance and azimuth [13]. The above capability is a result of advancements in antenna design where the beam-forming enabled CRs to not only measure the power level in a particular band but also the direction from where the signal arrives [13]. In code dimension, channel sensing refers to the identification of spread spectrum signals which can be Frequency Hopping Spread Spectrum (FHSS) or Direct Sequence Spread Spectrum (DSSS) and finding out whether there exists possibility to use an orthogonal code to transmit secondary payload [13].

As one can see, CR spectrum sensing require an antenna with a wide operating range, analog to digital converters with sampling frequencies in the range of GHz with very high resolution, low noise power amplifiers with a wide dynamic range and digital signal processors with very high operating speeds [13]. Another challenge in spectrum sensing is the hidden terminal problem. This happens due to the PU signal on a sensed channel being undetectable at the CR receiver due to shadowing, severe multipath fading or high penetration loss of a building [14]. In this case although a transmission by the CR may not interfere with the primary transmitter, it might interfere with the primary receiver. As one can see, the propagation path between primary transmitter and cognitive radio transmitter which senses the channel is independent of the path between primary transmitter and receiver. Therefore a CR not only has to have a receiver which is 30-40dB more sensitive than the primary receiver but also has to decide whether the channel is idle or busy looking at the measurements using the indirect path [15]. These are the main challenges on spectrum sensing faced by researchers these days.

In channel sensing we test whether the PU is present or not in the sensed channel. Therefore it boils down to a binary hypothesis test problem, where the two hypotheses are given as [14]:

$$\begin{aligned} \mathcal{H}_1 &: \text{ PU is using the channel} \\ \mathcal{H}_0 &: \text{ PU is not using the channel} \end{aligned}$$

In this hypotheses test, the effectiveness of the sensing method can be measured by the probability of false alarm P_f and probability of mis-detection P_m. False alarm: mistakenly concluding the channel is busy with PU traffic when it is not. Mis-detection: concluding the channel is idle when PU is operating. These imperfections in channel sensing can be mathematically expressed as [14]:

$$P_f = Prob\{Decision = \mathcal{H}_1|\mathcal{H}_0\}, \tag{1.3.1}$$

$$P_m = Prob\{Decision = \mathcal{H}_0|\mathcal{H}_1\}, \tag{1.3.2}$$

In the literature there have been many efforts to combat the challenges highlighted above. The methods in the literature introduced for channel sensing include Energy detector based sensing, Waveform based sensing, Cyclostationarity based sensing and matched filtering [13]. The Energy detector based sensing is considered to be the least complex of the methods but can lead to poor performance when the channel noise is non-stationary. The waveform based sensing has comparable performance with any of the other methods with complexity being in the mid range, but the primary user signal patters need to be known a priory. We do not delve into much details about these methods as sensing methods are not the main focus of the proposed work. Interested readers are referred to [13] and the references therein.

The next challenge introduced above was the problem of hidden terminal. In the literature many methods to combat this problem have been introduced which are based on the cooperation between the SUs. In the cooperative sensing regime each SU sends either its decision on the channel occupancy or the metric SUs use for the detection to either a centralized fixed location [16] or another SU [17]. The case where the decisions exchanged are used for deciding whether the channel is occupied or not is called hard decision fusion and the case where the metric is exchanged is called the soft decision fusion. The complexities associated with soft decision fusion involves requiring a higher bandwidth and complex processing at the fusion center. The centralized and decentralized approaches have the same pros and cons which are explained in the next section.

The scheme proposed in this thesis is not dependent on the sensing method used, but due to the simplicity and less PU signal information requirement we choose Energy detection as our sensing method. The motivation behind our proposed scheme is not about improving a physical sensing technique, it is rather about the management of channel sensing such that the SUs sense the channels which are more likely to be idle. To keep the sensing result sufficiently accurate while having less communication burden, we reduce the cooperation of SUs to only the transmitter and the receiver with OR-hard decision fusion. In OR-decision fusion the channel is considered to be idle only in the event that both SUs sense it as idle otherwise it is considered to be busy.

In the next section we focus on several spectrum sharing models used in the literature to make the reader aware of the reasons behind the network setting we are using.

1.4 Different Spectrum Sharing Models in Cognitive Radio

The spectrum sharing models in CR networks can be differentiated according to the network architecture used, adopted spectrum allocation behavior and spectrum access technique used [6]. The two main categories of network architectures in CR networks are networks with centralized control and networks with distributed control. They can be further categorized according to the spectrum allocation behavior as Cooperative and Non-cooperative. Finally, they can be categorized as Overlay and Underlay according to the spectrum access technique. These categories are further explained in what follows in this section.

1.4.1 Centralized Network

In the centralized network setting, decisions on channel assignment are taken by a secondary base station or a central spectrum server [18]. In this architecture, the base station gathers the spectrum sensing information from the SUs or a dedicated sensor network. Using this information it runs an assignment algorithm and informs the SUs of the channels to be utilized using a predefined control channel [18]. Having information on the global setting, the base station can provide a globally optimal channel assignment but there are several problems that are inherent to this architecture: requirement of a common control channel and high processing complexity at the base station [18]. In this type of networks common control channel should be pre-assigned and all SUs should have interference free access to it. In CR networks, such a channel may not exist although it is assumed in the literature and although a channel exists it will be highly congested.

1.4.2 Decentralized Networks

Decentralized networks are preferred to centralized when there is no preexisting infrastructure available or when the centralized networks are not scalable [6]. CRs use distributed algorithms in performing spectrum assignment in the decentralized setting. Each user does its own channel sensing and coordinates with the nearby cognitive radios and determines who gets which channel [18]. These algorithms are iterative so the neighborhood to which the CR sends data and from which it receives data should be limited. Otherwise it can incur large delays and the network will become unscalable [18]. The decentralized networks are also not completely immune to the problem of common control channel because to communicate with one another transmitter and the receiver should perform a handshake and it should happen on a channel both transmitter and the receiver know about.

1.4.3 Cooperative and Non-cooperative Networks

All the centralized CR networks are cooperative networks in general and some decentralized networks can be cooperative as well [6]. In [19] the authors formulate Game theoretic models for both Non-cooperative and Cooperative networks. The choice of the utility function plays a main part in defining a network as cooperative and non-cooperative. Each user tries to maximize the individual gain it can achieve without considering about the utility of others in non-cooperative networks. In contrast to that a user in a cooperative network tries to maximize its own utility while satisfying the minimum utility required by other users in the network. One advantage of non-cooperative networks over the cooperative ones stems from the advantage of being able to function with minimum communication with other CR users. The Cooperative networks can improve the overall network utilization with the burden of higher communication overhead. In situations where the benefits of cooperation surpasses the cost of communication one should choose a cooperative solution.

In the next section we discuss several applications of CR to emphasize the importance of this technology.

1.5 Applications of Cognitive Radios

In the IEEE 802.22 standard CRs are considered as a solution to provide rural areas with broadband Internet. These networks are called Wireless Regional Area Networks (WRAN). WRANs are centralized networks with an area of coverage having a radius of 33 km [20]. The FCC has released VHF and the UHF spectrum bands for the purpose of making rules for this standard. These bands are chosen because the broadcast television spectrum is very sparsely used and it has very good propagation characteristics. The PUs who should be protected in these frequency bands are the television broadcasters and viewers, Wireless microphone systems in concerts, private land mobile radio services and commercial mobile radio services [20]. In the 802.22 standard, the minimum planned downstream data rate is 1.5 Mbps and the upstream peak throughput is 384 kbps [20].

Another application of CR is a military application called joint tactical radio system which is considered as the backbone of the US Army's proposed future combat system. Originating in mid 1990s the main aim of this system program was to replace the 25-30 families of different radio systems with software defined radios which can operate in the entire frequency spectrum [21]. These software defined radios should also be able to communicate with the legacy radios used by the department of defense. The main purpose of the joint tactical radio system is enabling the seamless connectivity to all levels of command and providing direct access to airborne and battle field sensors [21]. Since this envisioned combat system overwhelms currently owned spectrum by the department of defense they will have to use dynamic spectrum allocation. One of the cognitive radio applications in joint tactical radio system is given in [22]. This application tries to adapt the radio band usage from air to ground based on the GPS coordinates of the location of the aircraft so that it will not cause interference to communications of ground troops.

Another area where FCC proposed the use of CR concepts is in the area of public safety communication [23]. This initiative was taken to provide the first respondents with better connectivity in emergency situations. Usually in this kind of scenarios the radio spectrum usage becomes concentrated both in space and in time since all the emergency workers will have no alternative communication methods other than the wireless radios [24]. Using CRs unlike the legacy radios one can avoid using the congested radio bands, where in legacy radios the band of operation is fixed. In addition the CRs can communicate with the legacy radios because of its advanced capabilities to operate in diverse frequency bands and standards. Another issue in public safety bands is they are scattered in a large frequency band and owned by different authorities. Thus the strict allocation policies create artificial spectrum scarcity in emergency situations and cognitive radio can be a solution to those situations [24].

In the next section we discuss the importance of Channel state prediction which is the enabling mechanism for the decentralized proactive channel access scheme studied in this thesis. We discuss the schemes found in the literature which uses proactive channel access in detail in Chapter 2.

1.6 Importance of Channel State Prediction

The channel access in CR can be classified into two broad categories: reactive channel access and proactive channel access. Most of the work on CR channel access have been done on reactive sensing and only a few publications in the literature focused on proactive channel access, which we will discuss in Chapter 2. An even lesser number of proactive access schemes merge a decentralized Medium access protocol together with a proactive channel access scheme. The main contribution of our work is this merging of proactive spectrum access with a decentralized MAC protocol.

In reactive spectrum access the SUs only sense the spectrum when they have some payload to deliver to another SU or receive from another SU. Therefore if the SU can only perform narrow band sensing, it has to either sequentially or randomly sense channels until an idle channel is found [25]. It is difficult to gather statistical knowledge about the channels out of this sporadic channel sensing [25]. On the other hand the data gathered by regular channel sensing can be used to create a statistical model of the PU channel usage for each licensed channel. The SU can make use of this model to find out the most optimal channel sensing pattern and intelligently schedule channel access in advance [4, 25]. Building this statistical channel usage model and using it to schedule channel sensing is called proactive channel sensing or channel state prediction.

In reactive spectrum sensing the sense and react policy of the SUs will unavoidably disrupt the communications of both the SUs and PUs, since the appearance of a PU within the time period between sensing slots is undetectable to the SU [4]. Because of the unplanned interruptions due to the PU user returning in the reactive sensing policy, the SUs cannot meet any quality of service requirements and they do not posses any knowledge of how long the current idle slot is going to last on average. On the other hand, the knowledge on the channel behavior which is gained by regular channel sensing and model formulation can be used to predict the effective bandwidth or the probability of the channel being idle in the next time slot, which allows the cognitive radio to adjust data rates accordingly [26].

1.7 SIC Techniques

The self interference signal received by a FD device consists of the signal power directly leaking from the transmit chain to the receiver chain and the signal power reflected off the environment. According to [27] this second type of interference is the hardest to deal with.

Therefore self interference cancellation is done in several stages namely: passive suppression at the antenna, analog cancellation at the radio unit and digital cancellation after analog to digital conversion [28]. Since there are pros and cons of all of these techniques they are used jointly to complement each other in self interference cancellation.

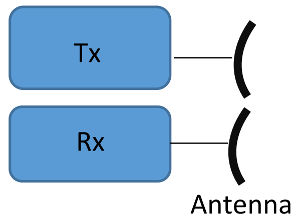
1.7.1 Passive Suppression

Passive suppression at the antennas can be achieved either by using directional receive and transmission antennas which have minimum intersection between the main radiation lobes or increasing the distance between the antennas [28]. There are two types of antenna configurations. The first one is separate-antenna FD and the second is shared-antenna FD [27]. A conceptual diagram of these two configurations are shown in figures 1.7.1a and 1.7.1b respectively. The suppression schemes depend on the configuration. The passive suppression leverages on isolating the transmit chain from the receive chain. Therefore in the separate antenna configuration this isolation can be achieved using path loss between the antennas [29], using different polarizations in the transmit and receive chains [30] and using directional antennas [31]. In the shared-antenna scenario the isolation is achieved using a circulator which is a ferrite device which shows non-linearity in the propagation of magnetic waves [27]. This passive suppression can only reduce the direct leakage of power from the transmitter chain to the receiver chain since the environment reflectors are dynamic.

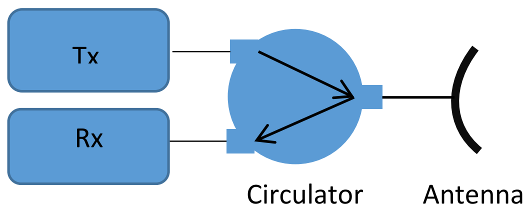
Experimental results in [32] reveal that up to $65dB$ of self interference can be canceled using omni-directional antennas and results in [31] show that using directional antennas up to $72dB$ cancellation is possible. The main problem with these isolators is their size.

1.7.2 Analog Cancellation

The concern that passive suppression techniques should be employed such that the desired signal is not suppressed motivated the analog cancellation. Furthermore, some amount of



(a) Separate-antenna



(b) Shared-antenna

Figure 1.7.1: Antenna Configurations for FD Transceivers

reflected transmit power can be canceled using analog cancellation. Here a gain, phase and delay adjusted copy of the transmitted signal is subtracted from the received signal at the antennas [27]. As explained in [33], there are two main methods to implement analog cancellation. The first is using either a balanced/unbalanced (Balun) transformer [34] or multiple parallel lines of preselected varying delays (i.e. wires of different length) and tunable attenuators to mimic the distortion the transmit signal goes through before receiving it in the same terminal and subtracting it from the received signal (eg: [35]). The second method changes the transmitted base band signal such that it includes all the distortion the transmit base band goes through before receiving at the receive antenna of the same device and uses a separate transmitter chain to create the analog cancellation signal (eg: [32]). This second method is more suitable for orthogonal frequency devision multiplexing (OFDM) type access.

1.7.3 Digital Cancellation

However, the above mentioned analog cancellation schemes also have their limitations. The first one is, the natural distortion the transmit signal undergoes before being received at the receiver antenna depends on the environment the device operates in. These environmental effects which are reflected in the channel state information estimation can be handled using training sequences in the digital domain more effectively [27]. Furthermore, constructing multiple delayed and attenuated versions of the same signal can be easily done digitally. However, in order for the digital cancellation to work, the self interference signal should not saturate the analog to digital converter (ADC). Therefore, the passive suppression as well as analog cancellation should precede the digital cancellation to bring the self interference to an acceptable level before the ADC. In this stage the linear and nonlinear residual self interference terms are canceled using digital signal processing [35].

1.8 Applications on FD communication

There are three FD topologies, which are listed below [36]:

1. Full duplex relaying
2. Bidirectional full duplex communication
3. Distributed full duplex (femto cells)

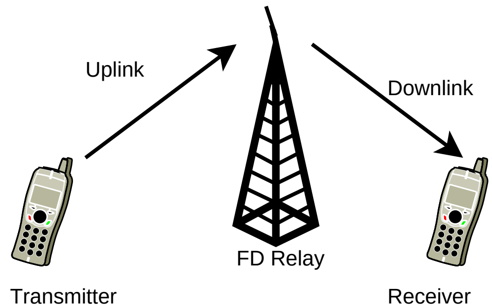
FD communication gives rise to two types of interference. One is self-interference the other is inter-terminal interference in relaying and distributed FD. Therefore, more care should be given when implementing FD protocols. Although FD increases the complexity of the design, it can be used to reduce protocol overhead by receiving the control information concurrently with the data transmission.

1.8.1 Full Duplex Relaying

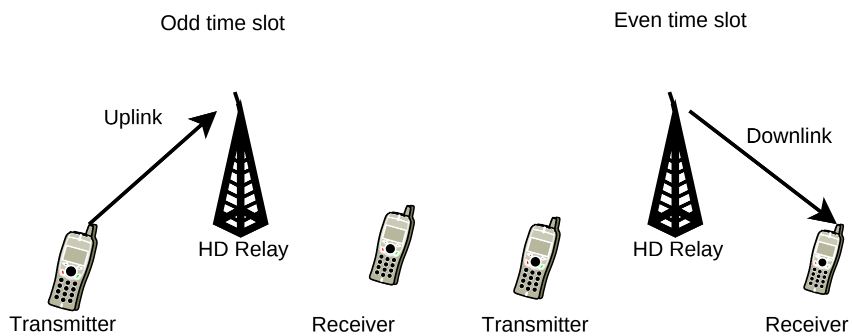
In full duplex relaying the cognitive base station or a cognitive user transmits the signal received by one party to another party concurrently. The relaying can both be amplify-and-forward relaying or decode-and-forward relaying (eg: [37]). In amplify-and-forward relaying only the passive and analog suppression of the self interference signal can be done. On the other hand in decode-and-forward relaying all three suppression techniques can be used. In FD relaying the relaying of packets from the source to the destination can be done in a single time slot whereas it takes two time slots for a half-duplex (HD) relay to do the same operation. A comparison between the HD relaying and FD relaying is shown in Figure 1.8.1. In Figure 1.8.1a the relay concurrently receives a signal from a source and transmits that signal to the intended destination. It is possible for the transmitted signal to be a delayed version of the received signal specially in the case of decode-and-transmit relays. In Figure 1.8.1b the reception and transmission happens on two time slots. For example the relay should finish receiving the entire signal before it can transmit it. Therefore the destination device and the source device should know there is a relaying device in place and synchronize with them accordingly. But in the case of FD relaying, the source and the destination need not know about the existence of a relay. Since the relayed signal is analogous to a time delayed version of the original signal which occurs as a result of multi-path propagation.

1.8.2 Bidirectional Full Duplex Communication

Here two devices which are close to each other can exchange data between the two concurrently. The decisions on the power and channels in these schemes can be decided by either a central entity or by the devices themselves in a distributed manner (ex: [38, 39]). If the system shown in Figure 1.8.2 is HD then either the link from node 1 to node 2 or 2 to 1 is active in a given time epoch.



(a) FD Relay



(b) HD Relay

Figure 1.8.1: Difference Between FD and HD Relaying

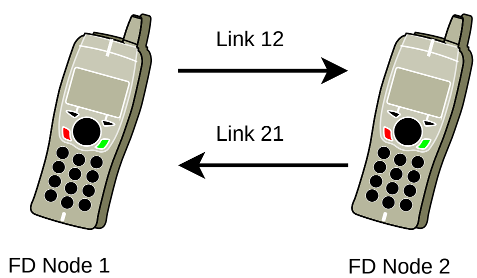


Figure 1.8.2: Bidirectional FD communication between two terminals

1.8.3 FD Cellular

In distributed full duplex the HD users are attached to a base station and the base station uses full duplex communication such that it receives data from one HD user on a particular channel while sending data to another HD user without having orthogonal channels for uplink and downlink. Then to reduce the interference from the transmitting HD user to the receiving HD user, the base station should select the pairs properly (eg: [40]). This scenario is shown in Figure 1.8.3 in the communication between the FD base station and HD nodes 1 and 2. The other scenario is where both the base station and the user equipment are both

FD enabled devices. This scenario is also shown in Figure 1.8.3 in the communication between FD node and the base station.

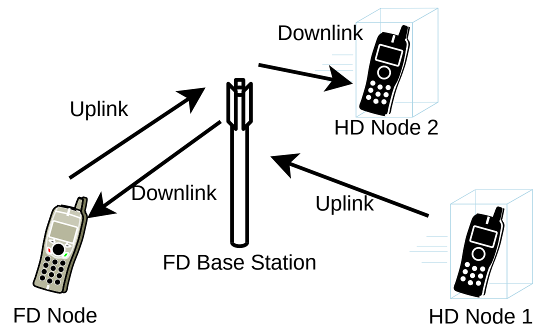


Figure 1.8.3: Data Communication in FD cellular Network

In the next section we present the outline of the report.

1.9 Outline of the Thesis

The main objective of this thesis is to identify methods of estimation and prediction to improve the efficiency of spectrum access by the Cognitive radios in both overlay and underlay networks. To this end in Chapter 3 we propose a Proactive Medium Access Control (PMAC) protocol for an overlay CR network which can be used to reduce the interference felt by the PUs while increasing the throughput of the SUs by having less channel search delay. Then in Chapter 4 we propose a FD underlay protocol for bi-directional CR transmissions with multiuser access and interference estimation.

1.9.1 PMAC Protocol for an Overlay CR Network

The work we are presenting on this topic tries to address some of the short falls in the literature. As we will see in Chapter 2 some of the schemes which give optimum results are too complex to be implemented online and some others which are based on learning have high learning complexities. We use a predictor which depends on the Markov properties observed in the channel state fluctuation patterns. The algorithm we use for prediction is called the probabilistic suffix tree (PST) algorithm.

In this scheme, we assume the presence of a radio environment map (REM) which collects channel usage statistics from a sensor network deployed in the geographic area of interest and from the cognitive radio users as in [41]. REM is an integrated database that contains information on services, networks, regulations and activity profile of radio devices in the network [42]. The REM is accessible to the SUs through a radio interface where the SUs use control channels to communicate with it. Before using the services of the REM, SUs should register themselves with this REM database and might be given some economic incentive for the activity monitoring operations they do for the REM. We assume unlicensed operation of SUs without any quality of service (QoS) guarantees, which is one of the two DTV band secondary spectrum usage schemes proposed in [43]. As the reader may see in Chapter 3, PST has a favorable property which enables us to calculate the probability of a channel being idle for channel state information strings having length 0 to a maximum length which is a parameter in the PST, which was one of the main reasons of using that modeling technique in this work.

A list of contributions to the literature presented in Chapter 2 are:

1. Proposing a single channel rendezvous scheme for the control phase which selects the channel hopping sequence based on the product of the stationary idle probability of the given channels and the mean idle duration of the channel. There are control channel establishment schemes which are based on single or multiple channel rendezvous like [44] and [45], but they do not consider the PU channel access statistics when selecting sensing and access sequences of channels.

2. Formulating an equation to calculate the upper bound of the number of control slots needed to successfully perform a Handshake between the SU transmitter and the receiver.

3. Formulating a marked Markov chain based method to select a channel set to be used in the Data phase of the communication between the SU transmitter and the receiver. As mentioned above this scheme reduces the channel sensing burden in proactive

channel access.

4. Using of Probabilistic Suffix Tree algorithm for channel sequence generation in the data phase of the Medium access protocol. We introduced the importance of this scheme in proactive channel access in [46], but in that paper we only considered the accuracy of prediction and complexity associated without considering about a MAC scheme.

The work reported in Chapter 2 was published in [47] and [48].

1.9.2 Underlay Protocol for Bi-directional FD CR Transmissions with Multiuser Access and Interference Estimation

Full duplex (FD) communication as well as Cognitive radio (CR) have both been proposed to increase spectrum efficiency [49]. Until the recent advances in self interference cancellation (SIC) techniques, the uplink and downlink transmissions between the transmitter and the receiver were either done in different time slots (Time division duplex) or different frequency bands (Frequency division duplex). According to [50] roughly $100dB$ isolation/cancellation between transmitter and the receiver is required for a satisfactory quality or service (QoS) in FD communication. Most work in the literature advocates the use of radio frequency (RF) isolation/cancellation at the antenna [31,32] and duplexer [30,32,35], and/or digital domain filtering of self interference [35, 51], to realize FD communication. For more information on SIC methods the readers are referred to the survey papers [33] and [27].

In this chapter we propose a bidirectional underlay FD CR network having monitoring stations (MSs) to measure the interference. The motivation in having the MSs is to get the information on the interference easily and to find the channel gain coefficients without much effort. Since the MS is a part of the secondary network we can design it to broadcast the control information on the SU interference observed. Furthermore MSs can broadcast training sequences periodically for the SUs to calculate the channel gain parameters. To get the PUs to do the same will require some form of cooperation between the secondary network and the primary network.

We approach this problem in a decentralized manner to reduce the control overhead and the computational complexity. A list of contributions to the literature presented in this chapter are:

1. Formulation of the problem as a decentralized joint channel assignment and power allocation problem.

2. Formulating the problem to have the flexibility to use some channels as FD and the others as HD so that the data rate between the pair is maximized.

3. Proposing an alternating optimization based solution method to this joint problem after decomposing it into channel assignment and power allocation.

4. Proposing a scheme based on least squares and Kalman filtering to estimate the level of interference at the MSs based on the local interference.

This thesis is organized as follows: Chapter 2 contains a literature review of the channel state prediction models found in the literature. In Chapter 3 we explain our channel selection scheme and the channel access scheme for an overlay network in detail. Then in Chapter 4 we propose a scheme for the channel assignment and power allocation based on estimation for an underlay CR network having MSs. Chapter 5 discusses the possible extensions to the work presented and concludes the report. In addition, Appendix A gives an overview of the probabilistic suffix tree algorithm and Appendix B gives an overview of the Monotonic optimization framework.

1.10 List of Publications

1. C. Devanarayana and A. Alfa, Proactive channel access in cognitive radio networks using statistical radio environment maps, EURASIP Journal on Wireless Communications and Networking, vol. 2015, no. 1, 2015.

[Online]. Available: http://dx.doi.org/10.1186/s13638-015-0309-2

2. T. O. Kim, C. N. Devanarayana, A. S. Alfa and B. D. Choi, ”An Optimal Admission Control Protocol for Heterogeneous Multicast Streaming Services,” in IEEE Transactions on Communications, vol. 63, no. 6, pp. 2346-2359, June 2015. doi: 10.1109/TCOMM.2015.2421896

3. C. Devanarayana and A. S. Alfa, Predictive Channel Access in Cognitive Radio Networks based on Variable order Markov Models, in GLOBECOM2011, 2011 IEEE Global Telecommunications Conference, 2011.

4. C. Devanarayana and A. S. Alfa, Proactive channel access in cognitive radio networks based on users statistics, in Cognitive Cellular Systems (CCS), 2014 1st International Workshop on, Sept 2014.

5. C. Devanarayana and A. S. Alfa, Decentralized Channel Assignment and Power Allocation in a Full-Duplex Cognitive Radio Network, in Consumer Communications & Networking Conference (CCNC), 2016 13th IEEE Annual, Jan 2016.

Chapter 2

Literature Survey on Existing Schemes

In the literature there are many prediction schemes proposed for half-duplex overlay networks but to the best of our knowledge no schemes have been proposed for its full-duplex counterpart. Furthermore interference estimation techniques have not been utilized for underlay cognitive radio networks to the best of our knowledge. In this survey we are presenting the prediction techniques used in half-duplex overlay cognitive radio networks first. Then we are presenting the rendezvous based control channel assignment schemes most of which did not used channel state prediction to get a general idea of what has been done already in the literature. After that we focus our attention on full-duplex schemes in the literature both underlay and overlay which are still to utilize prediction and estimation techniques.

2.1 Half-duplex Overlay Networks

In the literature there are proactive channel access schemes with Medium access protocols for both decentralized adhoc SU networks and centralized SU networks. In this section we discuss several of the current already published schemes and the differences in the proposed scheme in this thesis to that of those published already. Here, first we discuss some methods based on optimization methods like Markov Decision Process (MDP), Partially Observable

Markov Decision Process (POMDP) and several heuristics based schemes.

2.1.1 POMDP Based Schemes

POMDP framework is an analytic framework that can be used to optimize a sequence of actions that can be taken in a system where the system changes its state from one to the other based on the previous state and action taken with some probability. It is known as partially observable since the true state of the system is unknown to the decision maker but an observation on the system gives some probabilistic mapping from the observation to the possible set of system states. This mapping distribution is known to the decision maker. In the setting of cognitive radio the true state of the system is the knowledge on the channel occupancy of the PU on all N channels at the time of the decision but due to the sensing imperfections such as false alarm and mis-detection and the impracticality of sensing of all the channels makes the system state partially observable. Zhao et al. in [52] and Chen et al. in [53] developed two schemes based on the POMDP framework. Both of these schemes advocate the importance of the joint design of the sensor and the medium access. The rationale behind this thinking is if we want a better sensing result we have to sense the channel for a longer duration which reduces the time we can utilize the channel for actual transmitting. But how much better we could do in sensing depends on the maximum interference we cause on the PU. The authors admit that the search for optimal strategy can be done only for a smaller number of channel because of the exponential growth of the complexity and it can only be done offline. Thus if the channel state transition probabilities varies with time it might not be feasible to recalculate the optimum policies. According to [54] the complexity of POMDP's are PSPACE-Complete therefore even less likely to be solved in polynomial time than NP-Complete problems.

In [52] after the formulation of the problem as a POMDP they formulate a sub-optimal greedy channel selection and access scheme which is based on a vector $\Omega(t) = [\omega_1(t), \cdots, \omega_N(t)]$, where $\omega_i(t)$ represents the probability of channel i being idle at time t. They assume the channels to evolve independently and each channel's behavior is assumed to be distributed according to a first order Markov distribution. The Markov distribution of each channel is assumed to be known to each SU. Then the channel having the maximum product between the bandwidth and the probability of being idle in the current time slot is selected for communication between the transmitter and the receiver. They develop an equation to calculate the $\Omega(t)$ in both transmitter and the receiver without any information transfer between the two which is based on the acknowledgement and the channel chosen to transmit. In that paper the authors assume that each receiver has a set of channels which it monitors regularly and these channels are known to the transmitter to form the initial handshake. They did not discuss the initialization of the vector $\Omega(t)$ when a new transmission starts but it can be assumed to be the stationary probability of each channel being idle.

Yunxia et al. in [53] also discuss finding the optimal sensing and access strategy based on a POMDP framework. The specialty in this case is that they incorporate the selection of sensor operating point in an energy detector into the decision framework. In other words they select: the threshold for energy received in a given channel to identify it as busy and the duration for which the energy is accumulated. These two parameters in turn dictate the probability of false alarm and mis-detection in the energy detector. Incorporating the operator characteristics changes the problem structure from POMDP to constrained POMDP, where the constraint comes from the maximum allowed interference on the PU. According to the authors only randomized policies are optimum solutions to constrained POMDPs which makes the solution computationally prohibitive. Then for the single channel case they propose a Separation principle where the optimum sensor operating point detection and optimum probability of transmission are separated from the optimum channel sensing policy. The formulation for the identification of the optimum sensor operating point, $(\epsilon_a^*, \delta_a^*)$ and optimum probability of transmission, $(f_a^*(0), f_a^*(1))$ are given in Equation (2.1.1), where ϵ_a^*, δ_a^*, $f_a^*(0)$ and $f_a^*(1)$ are optimum probability of false alarm, mis-detection, access probability when channel detected as busy and access probability when channel detected as idle, on channel a respectively.

$$\begin{aligned} &\underset{(\epsilon_a,\delta_a),(f_a(0),f_a(1))}{\text{arg max}} && \epsilon_a f_a(0) + (1-\epsilon_a) f_a(1), \\ &\text{subject to} && (1-\delta_a) f_a(0) + \delta_a f_a(1) \leq \zeta. \end{aligned} \tag{2.1.1}$$

We believe that the left hand side of the constraint should be multiplied by the stationary probability of the channel being idle. Otherwise the interference caused on a given PU depends only on the access probabilities and the probability of mis-detection of the SU and independent on the channel usage by the PU. After the separation the optimum channel selection for sensing becomes an unconstrained POMDP because the interference constraint will be satisfied by the sensor design and access policy calculated in Equation (2.1.1) for any sensing policy. Then they extend their work to the case where the SU can sense multiple channels at once. Similar to the single channel case the joint design is computationally prohibitive. They mention that the separation principle does not hold in general but when the channels behave independently from each other the separation policy can be applied.

2.1.2 MDP Based Schemes

Because of the complexity in the policy calculation in POMDP framework Zhao et al. work around the problem of partial observability using periodic channel sensing in [55]. In this work they did not explicitly consider the probability of false alarm and mis-detection. As the authors mentioned this scheme is sub-optimal since the channel sensed to be idle at the start has a very small probability of being idle now. On the other hand a channel which was sensed busy will have a better reward since it has less probability of being busy at the current time slot. This happened because multiple transitions in between the sensed time and current time were not considered. In this work the busy period of channel i was considered to be distributed according to an exponential distribution with rate μ_i and the idle state was considered to be distributed with rate λ_i. Because of the non-time slotted operation the PU can occupy the channel at any time therefore a constraint is drawn to keep the interference to the PU below a threshold which made the problem a Constrained Markov

Decision process. For constrained processes only randomized policies were feasible which makes the policy calculation complex. Since the policies were periodic the problem was formulated to maximize the infinite horizon average reward.

Then the authors proposed two heuristics-based schemes: Memoryless Access and Greedy access. In the first scheme they carried out periodic sensing as in the CMDP scheme and if the channel was sensed idle the SU transmitted with some probability to ensure that the interference constraint was satisfied else it did not transmit. In the greedy access scheme, the channel having the maximum idle probability was chosen for transmission and transmission was done with some probability to ensure the interference criteria is met.

In the literature there are many heuristic based schemes employed for channel state prediction. These methods can be divided mainly into two categories: prediction schemes which model the channel usage patterns based on the past channel usage data and prediction schemes which assumed channel usage models for the licensed channels (eg: [56], [25] , [4]). A few examples of methods used in the first category of predictors are Hidden Markov Models, Neural networks, Regression techniques and Variable order Markov models (eg: [26], [57], [58] and [46]). In the next two subsections we are discussing these two types.

2.1.3 Prediction of Channel Status Using Past Channel Usage Data

In this subsection we discuss few of the methods used in channel usage modeling. These models enable the cognitive radios to predict the future channel states given the past channel states. Channel state in this context means whether the channel is occupied by a primary user or not. We set the channel state to *busy* when the channel is occupied by a licensed user and set it to *idle* when the channel is not occupied.

Prediction Using Hidden Markov Models

In [57], the authors discussed a scheme with multiple PUs and a single SU, where the SU used hidden Markov models to model the channel usage behavior of the PUs. In this scheme not much was discussed about the type of network or the MAC protocol used; in addition, the sensing imperfections were not taken into account. In a hidden Markov model the states of the Markov chain are not physically identifiable. The only observation that can be made is the symbol which is generated by a distribution conditioned on the current state the Markov chain. A Hidden Markov process is a bivariate stochastic process [59].

A discrete time hidden Markov model consists of a *N*-state Markov chain having a state set $\mathbf{Q} = \{q_1, q_2, ..., q_N\}$ and at each transition time epoch *t* the Markov chain enters into a new state based on a transition probability matrix $\mathbf{A}$ given the previous state. These states $\mathbf{Q}$ are not observable to the outside world. Then at that particular time instant a discrete observation output symbol $v_i \in \mathbf{V} = \{v_1, v_2, ...v_M\}$ is produced based on a probability distribution $\mathbf{B}$ which only depends on the current state of the Markov chain. For example $v_i \in \mathbf{V} = \{idle, busy\}$. These states $\mathbf{V}$ are observable. The general initial state distribution of the Markov chain is given by $\pi = (\pi_1, \pi_2, ..., \pi_N)$ which can be the steady state distribution or some other user defined distribution [59]. In compact notation a hidden Markov model is usually expressed mathematically as $\lambda = (\mathbf{A}, \mathbf{B}, \pi)$.

The above mentioned parameters $\mathbf{A}$, $\mathbf{B}$ and π should be estimated based on a given string of past observation data. Baum-Welch algorithm was used to estimate the parameters in this scheme. A separate model has to be built for each channel if the channel usage is different statistically. If the model for a given channel j is $\lambda_j = (\mathbf{A}_j, \mathbf{B}_j, \pi_j)$ one can calculate the probability of the next slot being idle or busy given the model λ and the channel usage data. When using hidden Markov models one has to maintain a large number of past observations and the model parameter estimation techniques are highly complex [58].

Prediction Using Binary Time Series

Yarkan et al. in [58] modeled the state of the channel which is either idle or busy as a linear function of the past observations, and they used a sigmoid function to keep the resulting value in the interval $[0, 1]$. The sensing results were considered to be perfect in this scheme too. They only concentrated on the prediction of the state of the primary user without giving any prominence to the Medium access of the SU. The conditional success of the current trial was estimated using this time series of channel states. Let the current status of the channel be given by B_t, where $B_t = 0$ when the channel is idle and $B_t = 1$ when the channel is busy. Probability of the channel being busy in the next time slot conditioned on the information known up to time $t-1$, $\mathcal{F}_{t-1}$, can be given as $P_\beta(B_t = 1|\mathcal{F}_{t-1})$ [60]. The information up to time $t-1$, $\mathcal{F}_{t-1}$, can include the past data on the channel status and/or any other variable or a set of variables which is known to influence the channel status. Where $\beta = (\beta_1, \beta_2, ...\beta_d)$ denotes the model parameters and d is the order of the model. Using regression for some parameter vector β, the expectation of the channel state in the next time slot given the information available up to the time instant $t-1$ can be found.

To find the value of B_t the parameter vector β should be determined first. When this regression is done concurrently for R channels this β becomes a matrix which is known as the co-variance matrix which takes into account the inter-dependencies of the channels. This estimation is done using a matrix of d past observations of R channels. Yarkan et al. found the co-variance and the intercept vectors using genetic algorithm and neural networks and found the probability of the next slot being idle for R channels, given the history of those channels for the past d time slots. For more information about $VAR(d)$ models the reader is referred to [61].

In the multivariate binary regression model introduced in [58], a closed form solution was not possible for the intercept vector and the co-variance matrix since the function had to be transformed using a link function, which was non-linear. Therefore a numerical approach was taken. As one can see in this technique the process of finding out the coefficient matrices are complex and a globally optimum solution is not possible due to the higher dimensionality.

Prediction Using Multiple Layer Perceptron

The aim of the method discussed in [26] was to develop a model to predict the next channel observation conditioned on the observations of the current time slot and the past consecutive d time slots of the given channel. In this method the next channel outcome was predicted directly rather than giving a probability of it being idle or not. The sensing results were considered to be perfect in the simulations conducted. They also did not delve into how this scheme can be used in a network with multiple SUs. In this scheme a Multiple layer perceptron model which is an Artificial Neural network was trained for this purpose. In this model neurons are arranged in layers and they are connected using weighted lines. A multiple layer perceptron has an input layer, an output layer and between those layers it has some hidden layers. The models created in [26] had a non-linear relationship between the inputs and the output. In [26] the channel states idle and busy were given the values -1 and 1. To train the neural network the input and the respective output data were required. Those values were obtained by the binary time series of -1 and 1, taken by sampling the channel for a time period. The number of inputs in the multilayer perceptron is called the order of the neural network [26].

In the training process weights on each edge are optimized so that the output gives minimum error for the training sequence. This weight optimization was done in [26] using the Bach propagation algorithm. Further information with regard to the Bach propagation algorithm can be found in [26].

In the multilayer perceptron technique the number of layers and the neurons per layer depends on the application [26]. Thus for different traffic patterns different architectures are required. The complexity of training is high for neural networks which can be a problem if the traffic patterns change depending on the time of the day.

2.1.4 Prediction Using an Assumed Channel Usage Model

In contrast to the earlier subsection, the models discussed here assume a distribution for the usage time of a channel by a PU. Then based on this distribution and the sensing result of the current time slot, one can predict the probability of the channel being idle in the next time slot. In this section we discuss five types of assumed channel models such as independent Erlang distributed alternating busy and idle times [25], independent exponentially distributed alternating busy and idle times [25, 56] alternating-periodic exponential model [4], biased geometric PU packet inter-arrival time [62] and Pareto distributed PU packet inter-arrival time [62].

In the models mentioned above expressions are derived for the probability of a particular channel being idle, given the sensing result of the past time slot. The expressions are derived using the theory of alternating renewal processes for both the Erlang distributed case and exponentially distributed case. The channel alternates between busy and idle states for these two cases and the duration for which the channel is in either of the states is statistically governed by Erlang distribution or the exponential distribution respectively. The busy and idle duration have different rate parameters and they are assumed to be independent of each other. The period of idle (busy) is exponentially distributed with parameter λ while the busy (idle) duration is fixed at a constant T in the alternating-periodic exponential model [4].

Gu et al. presented a centralized proactive channel hopping dynamic spectrum access method where the time duration between two sensing operations across the channels are different in [56]. They assumed the presence of a common control channel so that SUs can be informed on the channel to access by the base station. They assumed that the base station has the current knowledge of the channel states of all the channels. In this work only one SU is considered to be sending data to the base station at a given time. The idle and busy time durations were considered to be distributed according to the exponential distribution. When the probability of return of PU on channel i becomes larger than that of channel j a channel switching is done from channel i to j inside the time slot. They did not consider channel sensing imperfections in this paper. They derived expressions for the theoretic average interference time and the upper bound of average delay for packet transmissions. They assumed each data packet takes time T for transmission and broke each packet into L sub-packets. Then each sub-packet consisted of mini-packets of arbitrarily small duration Δt. They found the optimum channel switching times and channels to be used such that the average interference time on delivering all sub-packets is minimized. This optimization was based on dynamic programming and the complexity rose exponentially with the number of channels. It is doubtful whether this algorithms can be executed online as proposed.

Song et al. proposed a decentralized proactive spectrum handoff framework for CRs where the presence of a common control channel is not assumed in [62]. They used the same idea as us where a channel hopping sequence is assumed for both control message and data message transmission. They assumed each SU to be equipped with two radios where one was used for control and data message exchange while the other continuously scanned a PU channel. In that work, they based their channel access and selection criteria on two parameters one was the probability of the channel being idle at the proposed time of transmission and the other was the probability of it being idle for the frame duration. Those parameters were calculated assuming a distribution for PU packet inter-arrival time and minimum packet length. In the distributed channel selection algorithm, they assumed each SU can generate the same pseudo-random channel selecting sequence where each SU claimed the channels based in this order. Since an SU only knew the time the most recent PU transmission ended and the minimum packet length of the PU channel it sensed, each SU who were transmitting in the next time slot broadcasted their neighbors about their own information and the information they gathered from their neighbors. Then based on the parameters for selection they arranged the channels and each transmitting SU claimed the channels in this random selection order. In the simulations they used two distributions for the packet inter-arrival time one is a biased geometric distribution and the other is the Pareto distribution. In this work they did not consider channel sensing imperfections or the feasibility of broadcasting PU channel usage information.

The methods discussed above assumed that the channel usage behaves statistically according to a particular distribution. Exponential distribution is a memoryless distribution. Therefore the residual time does not depend on how long the process has been going on. Field studies done on the sojourn time of idle and busy times have shown that this assumption of exponentially distributed sojourn times of channel states cannot be taken for granted [63]. Therefore to find the closest match for the sojourn times a distribution fitting must be done. Out of the distributions used above, only the Pareto distribution gives a closer match to the actual scenario due to the fact that most data networks have self similarity [64].

In the next section we discuss some common control channel design techniques found in the literature.

2.2 Existing Common Control Channel Designs

The CCCs proposed for overlay networks can be divided into three main categories [65]:

1. Sequence hopping based CCC
2. Group based CCC [45]
3. Dedicated CCC

There are a few publications on the rendezvous based control channel assignment schemes in the literature. However all of these schemes focus only on the control channel and do not give any results on data transmissions together with rendezvous based control channel.

Lo et al. discussed a control channel assignment scheme for efficient recovery for a CR Ad Hoc network in [45]. In that scheme, they did not consider any statistical properties of the PU channels. They made a sequence of the potential control channels based on the signal power present on the channel. In this sequence, they arranged channels having signal power less than the PU activity detection threshold from the lowest power to the highest. Then based on this order, they randomly picked channels to make a channel sequence of length n. In this sequence, the channels having lower signal power are given higher probability of being selected. So there were multiple occurrences of the same channel in the sequence. Then they transmitted beacon messages on channels according to this channel hopping sequence based on a CSMA based protocol. Then if a neighbor's channel sequence coincided with the transmission, those receivers replied with an ACK message. Afterwards, each channel was given a weight according to the number of ACKs received. After dwelling in the discovery phase for a predetermined time period, a single control channel or multiple channels were picked based on the weights. Once a previously idle PU channel becomes busy, this channel gets removed from the control channels. Then the next best channel in terms of weight was used for the time being until the discovery phase is activated again. As one can see in this method the decision on control channel depends on the instantaneous signal power and quorum between the neighbors. The burden of discovery can be high when the PU channel status changes fast.

In [66], the authors discussed several channel sequence arrangement methods to be used in control channel blind-rendezvous schemes and found out the expected time to rendezvous for each of them. But in this analysis they did not consider a multiple CR scenario. Furthermore PU channel usage statistic were not considered in any of the schemes. They assumed that the control channel sequences contained available PU channels, but the methods to find the availability of channels were not discussed.

Bian et al. [44] discussed channel hopping sequences based on Quorum systems. Given a universal set U having integers from 0 to $n-1$ a quorum system S is a collection of subsets of U, where any two subsets p and q belonging to S have an overlapping of at least one element, in other words $p \cap q \neq \emptyset,\ \forall p, q \in S$. By assigning time slot indices and frame indices in a systematic way to each subset in S, they found quorum based channel hopping sequences to be used in the control phase. The probabilities with which each channel becomes idle have not been taken into consideration in these channel hopping sequences.

2.3 Full-duplex Cognitive Radio Protocols

Full-duplex (FD) cognitive radio (CR) protocols can be divided into two main categories depending on the way they access the spectrum. These categories are overlay protocols and underlay protocols. These two methods were discussed in Section 1.2. Both of these spectrum access methods can be used in bi-directional FD, FD relays and FD cellular networks. We will discuss some of the schemes in the literature on FD CR protocols briefly in the next two sections.

2.3.1 Underlay Protocols

Full-duplex Relaying Networks

In [67], the authors discussed the information theoretic rate region of a network where there are two CRs and two primary users (PUs). A rate region is the region inside of the plot of the Shannon's capacity of the rate achievable by the primary system against the rate of the secondary system. Here the primary transmitter transmitted data to the primary receiver. Then the secondary transmitter concurrently relayed the primary signal and transmitted the data it wants to transmit to the secondary receiver. In this transmission by the secondary transmitter it used a fraction θ of its total power to relay the PU signal and the remainder was used for its own transmission. The idea here was that, the interference the secondary user (SU) transmitter caused at the primary receiver can be compensated with the extra signal boost given by the SU by relaying the PU signal. Here the secondary transmitter acts as a FD transceiver since it transmits both its own signal and the delayed primary signal while receiving the primary signal to be relayed. Perfect self interference cancellation was assumed at the secondary transmitter. Then developing this concept further with the incorporation of residual self interference, Zheng et al. [37] derived the rate region for a cognitive base station which relayed primary user signals and transmitted data to cognitive users while receiving the primary signal to be relayed. In this scheme also, they considered a primary base station and a single primary user in the primary system. In the secondary system there was a cognitive base station (CBS) having N_t transmitting antennas and N_r receiving antennas and a single CR. The CBS is the only node capable of FD communication. The receiver antennas receive the interfering signal from the transmitting antennas in addition to the primary signal. In this scheme both the amplify-and-forward and decode-and-forward schemes were considered. As the names suggest, in the amplify-and-forward scheme the CBS amplifies the received primary signal and send it. In the decode-and-forward scheme the CBS decodes the primary message first and then transmits this signal. Then for a given interference suppression matrix between the transmit and receive antennas at CBS, Zheng et al. found the beam-forming matrices to transmit the primary signal and the secondary signal such that the rate achieved by the CBS was maximized while guaranteeing a predefined rate for the primary user and having a total power constraint for the CBS. They showed that the rate region could be significantly enhanced by using FD relaying. Furthermore, they showed that a scheme cooperating with the primary system provides a larger rate region than a system where the CRs used beam-forming to nullify the interference at the PU.

Full-duplex Cellular Networks

In the CR scenario FD is used in small cell networks which can be either one of femto or pico cells. A small cell is a low power base station serving a limited number of users within a small geographic area using the same frequency bands as the macro base station, which covers a larger area including the area covered by the small cell. These small cells cover the areas the macro base station reception is hindered. Furthermore these small cells increase the achievable data rates of the users since the distance from a small cell to a user is far less than that of a macro cell. This reduction in distance enables the users to enjoy higher signal strength and thereby higher data rate. These are part of a self organizing network standardized in the third generation partnership project (3GPP) release 9 [68] to improve the capacity and coverage [69].

A duplex mode selection algorithm based on stable roommate matching for a full duplex cognitive femto cell network was proposed in [40]. In stable room mate matching, a set of cardinality n, which is even, is partitioned into $n/2$ pairs. In these pairs there cannot be any two people who are not a pair prefer each other to their own partners [70]. In [40], Feng et al. assumed that the power allocation is fixed in both the macro-cell transmissions and femto cell transmissions. Since the femto cell used the same frequency bands as the macro cell the macro cell user was considered as the PU while considering the femto cell users (FUs) as SUs. Here, a single PU equipment served by the macro base station using orthogonal frequency division multiplexed access (OFDMA) was considered. This OFDMA protocol used N channels each having a bandwidth W. The cognitive femto base-station was assumed to be the only entity that had the capacity to perform full-duplex transmissions. Their optimization problem consisted of selecting the pairs of FUs for uplink (UL) and downlink (DL) transmissions such that the resulting SINR of the macro user, macro base station, femto base station and the FUs belonging to F femto base stations were above given thresholds. First, channels were allocated to FUs for UL and DL transmissions using the stable roommate matching algorithm without considering the inter femto cell interference. Thus this was run independently at each of the F femto cells.Then a greedy algorithm was used to allocate channels to femto base stations such that the rate was maximized and the SINR thresholds were not violated. In case no feasible pair could be found for UL and DL both (i.e. FD), the femto base station used one of UL or DL that maximizes the femto-cell throughput (i.e. HD).

Bi-directional Full-Duplex Networks

In [49], a power allocation scheme for a FD underlay CR network was developed using a proportional-integral-derivative (PID) controller. In this scheme, stations which monitor the SU interference on the PU channel was introduced. These stations were called detection points (DPs). Since this was an entity in the secondary network, reporting of the interference back to SUs could be done easily. Furthermore, the DPs could send control sequences to determine the channel gains. The challenge in this type of a network is finding the acceptable level of interference at the DPs such that the PUs interference thresholds are not violated. In this scheme the authors only considered a single channel. The main objective was to keep the SU SINR above a given threshold while keeping the interference at the DPs below a given threshold. In our scheme we used a network setup which is same as this scheme.

A power allocation scheme for the high signal to interference plus noise ratio regime was discussed in [71]. However they only considered the case of having a single SU. Therefore they did not take into account the interference from other SU pairs. Not considering the inter SU pair interference makes this solution less practical. Only a single PU was considered in [71]. In this scheme the goal was to maximize the sum throughput of the SUs while limiting the probability of outage of the PU. They converted the non-convex problem into a convex one by approximating $\log(1 + SINR)$ with $\log(SINR)$ (i.e. assuming high SINR) and then using a variable substitution. Then they found the optimal solution using Lagrangian relaxation and the Krush-Kuhn-Tucker conditions. In the result calculation they found the self interference cancellation factor ranges for which the HD and FD is optimal. Here they assigned one node of the pair as master and the other as the slave and consider the flow from master to slave only.

2.3.2 Overlay Protocols

In overlay protocols, the term full-duplex is used to identify a scenario where the CRs can sense the channel while transmitting. Therefore the actual data transmission operation is still half duplex. They do not fall into the full duplex categories mentioned above.

In [72], the authors presented a FD sensing scheme and a FD medium access control (MAC) scheme for an ad-hoc CR network, where there was no synchronization between the CRs and their primary counterparts. Both PUs and SUs were assumed to be using a p-persistent CSMA (collision sense multiple access) protocol for transmission, while the PUs had priority over the SUs. Their main motivation was to minimize the reactivation failure problem for the PUs in this non-time-slotted protocol. Reactivation failure problem is the incapacity of SUs to detect PU activity after an idle period. Sensing at the start of the time slot could not guarantee the PU was inactive, since there was no synchronization between the PUs and CRs. Sensing at the start of the time slot has been the norm of overlay half duplex communication. Their protocol used the ability of the FD transceivers to sense and transmit at the same time to ensure that primary user transmissions are not interfered.

The problem of optimal selection of the lengths of the sensing slot and the data transmission slot in a FD transmit and sense (TS) scheme and, the initial sensing slot length and the data transmission slot length of a transmit and receive (TR) scheme were discussed in [71]. In this scheme the authors assumed the distributions of the PU busy time and the idle time were known a priory. The sensing and transmission time durations were found based on a brute force search. Furthermore the authors found the ranges of probability of PU being idle in which the TS scheme is optimum and the TR scheme is optimum.

The probabilities of false alarm and mis-detection in both listen-before-talk (LBT) and listen-and-talk (LAT) protocols were derived for a pair of SUs with two antennas and a single PU in [73]. Furthermore, they proposed a switching protocol to choose between the LBT and LAT protocols. This switching was based on calculating throughputs for both LBT and LAT protocols and selecting the larger one.

In [74], a CSMA RTS/CTS medium access control protocol for a FD cognitive radio is proposed . In this protocol they fragmented the SU data packets such that the packet transmission time of a single packet is less than the channel evacuation time. They assumed exponentially distributed active and idle times for the PU and found an expression for the average throughput of the system based on the parameters maximum back-off interval, the fragment length and the transmit power. Then they found the optimum transmit power for each value of fragment length and back-off time. Finally they found the best combination in an exhaustive search.

In [75], the authors discussed a centralized cognitive radio network, where the secondary base station used the receiving antenna to sense the spectrum and the transmitting antenna to transmit to SUs concurrently. The authors showed that although the SU throughput gets better with high transmit power the sensing operation gets worsened. Therefore they argued that power control is essential in FD overlay schemes. The authors incorporated the power allocation into the overlay network, which is different from above overlay schemes. Here the authors assumed there were multiple PU channels and only one channel was associated with one user. Then they solved the channel-SU matching and the power allocation problem approximately using a 2-dimensional matching algorithm, while iteratively adding power to the best link. In other words the power was equally divided into a number of discrete units and at each iteration one unit was added to the best link. In the context of [75], a link is a matching between a channel and a SU.

2.4 Applications of Device-to-Device (D2D) Communications

D2D communication protocols have been proposed to augment the cellular communications in LTE Release 12 [76] and the releases thereafter. The 3GPP standardization body is expecting to increase the throughput of the Long-Term Evolution (LTE) wireless networks thousand fold increasing the spectral efficiency, cell density and the spectrum used. LTE belongs to the fourth generation of mobile networks, although it is still evolving the research and the industry are envisioning a newer type of mobile network which belongs to the fifth generation of mobile networks. This new generation of networks is supposed to have many features such as Enhanced mobile broadband, Internet of everything, Ultra reliable communications and very low latency [77]. The future mobile traffic in 2020 is expected to produce 8K ultra high definition video data which is 2600 fold larger than the entire traffic of 2010 and will connect more than 50 billion devices engaging in machine-to-machine type communications according to [77]. Since the 5G networks are expected to support connection of devices like sensors which generate very low amount of data while being constrained of the available power, it must be extremely power efficient and cost effective [78].

To support this enormous increase in mobile traffic many new techniques should be incorporated in wireless networks. Some of these ideas include using D2D communication, using smaller cells, usage of new spectrum in the 20-80 GHz range (mmW), using advanced multi-antenna systems, using flexible spectrum usage schemes like cognitive radio and using flexible duplex schemes [77].

D2D communication can be used in the following categories of services [79]:

1. Local services

2. Emergency communication

3. Internet-of-things (IoT) enhancement

2.4.1 Location based services

Local services that can be offered using D2D communication includes playing games with users nearby and exchanging data between users without going through the base station, advertising products based on proximity and setting up D2D based local media services. Using this third application the mobile operators does not have to stress their network to enable the users to download HD content in their mobile phones.

2.4.2 Emergency communication

Usually in situations of natural disasters the networking infrastructure gets damaged or loses power from the grid. This makes the mobile devices disconnected and unusable. D2D communications can play a major role in connecting users in scenarios like these. The devices can form a multi-hop ad-hoc network and connect devices together, and the mobile devices which have cellular coverage or a connection to the intended recipient can deliver the message.

2.4.3 IoT enhancement

As explained in [79] one of the possible use cases of IoT is in vehicle-to-vehicle networks, where a vehicle can inform the nearby vehicles before it makes an abrupt change in the speed or does a lane change while traveling at high speed. Another application is the aggregation of data generated by sensors in a smart home or a smart grid and sending it to the cellular network via a single special mobile terminal. This special terminal connects to other low-cost sensor terminals via D2D communication and after collecting the data sends it to the cellular network. This alleviate the cellular network from having to connect to all the low-cost sensor nodes.

2.5 Summary

In this chapter, first we discussed several channel assignment schemes designed for the control phase and data phase focusing on their pros and cons. Then we discussed some full-duplex cognitive radio protocols used in different network setups.

Although the schemes in [53] and [52] were capable of providing an optimal data rate for the SUs, the complexity of solving them are prohibitive for large number of channels and longer horizon lengths. The MDP based scheme discussed in [55] provided a sub-optimal solution to the problem of channel selection using a periodic sensing approach. This framework ignored the probability of false alarm and mis-detection. The complexity of solution methods provided in [55], [53] and [52] increases exponentially with the number of channels. In the learning based schemes mentioned in this chapter the learning complexity of MLP, HMM and binary regression techniques are higher than that of the PST scheme which we used. Furthermore, the MLP and binary regression based methods cannot be used in a scenario like ours, since they needed observation sequences of length equal to the order of the model. The schemes which assumed a distribution for the PU channel access did not discuss on how good those assumptions are in a realistic traffic scenario which shows bursty characteristics.

In our proposed scheme in Chapter 3, we filtered out the channels based on the channel usage models, and selected a set of channels such that they can be monitored regularly. Then we used a prediction scheme which is less complex, but it can perform almost as well as an HMM for the predictions. Unlike the other schemes we took the channel access of other SUs into account in filtering out the channels to be used in the data phase.

The full-duplex schemes in the literature did not use any form of interference estimation at the primary users. Furthermore, we proposed a joint channel assignment and power allocation scheme which can decide the channels to be used as half-duplex and the ones to be used as full-dupex in Chapter 4. The optimization scheme in [71] did not consider the inter SU interference, which was unrealistic. In [49], the cognitive radios only tried to achieve a pre-calculated data rate, which is acceptable if the quality of communication cannot be improved with extra data rate (eg: voice communication). Since it is based on feed back it might not be usable in a dynamic network. Furthermore, these schemes did not take into account the receiver sensitivity.

Chapter 3

Proactive MAC Protocol for an Overlay CR Network

In this chapter, we discuss our proposed scheme for proactive channel access in an overlay cognitive radio network. In this scheme, we assumed the presence of a centralized radio environment map (REM) which collects channel usage statistics from a sensor network deployed in the geographic area of interest and from the cognitive radio users similar to [41]. This REM is enhanced with statistical knowledge of the primary user (PU) channel usage, where a similar assumption was made in [80]. The SUs registered themselves with this REM and could be given some economic incentive for the sensing operations it does for the REM. We assumed unlicensed operation of SUs without any QoS guarantees, which was one of the two Digital TV band secondary spectrum usage schemes proposed in [43]. We assumed that the REM periodically created variable order Markov models (VMMs) using the probabilistic suffix tree (PST) algorithm and calculated the mean idle period lengths for a set of N channels and sells them to the SUs. More information on this PST algorithm is given in Section 3.2.1. Then based on the sensing outcome and the purchased channel usage model, the SUs calculate the probability of the channel being idle in the next time slot. Based on these probabilities, SUs rank the channels in the non-increasing order of availability. In this *Proactive Channel Ordering scheme*, the probability of a channel being idle is conditioned on the past consecutive sensing outcomes of that channel. Since the SU does not sense all the channels all the time, it does not have sensing information of some channels and have information of several consecutive past state information on some others. The main reason to use the PST algorithm in this scenario was the capability of creating a tree having conditional probabilities on varying conditioning strings. Because of this capability, we could calculate the probability of a channel being idle conditioned on channel state information strings having length 0 up to a maximum length d. This maximum length d is a parameter of the PST algorithm.

In the proposed scheme, we used a MAC protocol based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). This MAC protocol was used in our scheme because there is no central authority to coordinate access for the SUs in terms of time slot assignment in TDMA or code assignment in CDMA. In the proposed scheme, we did not assume any pre-existing control channels. Here, the control channel at a particular time slot was determined based on a channel hopping sequence consisting of the set of PU channels, which were also used for Data transmission. For this control channel assignment scheme we developed an equation to find the upper bound of the number of time slots needed to acquire a control channel for a pair of SUs.

According to Xing et al. in [81], all of the proactive spectrum schemes they surveyed did not take into account the effect of channel access of other SUs when selecting channels for the SU of interest. In addition, most heuristics based schemes do not base the channel selection on long term behavior, rather it is just based on selecting a better channel for the next time slot. Out of the channel selection schemes proposed in this thesis for the data phase, the first scheme only considers the effect of other SUs. In the second scheme, we calculate the reward of using each channel considering both the long term future behavior of PU channel access and the effect of channel access of other SUs. Using extensive simulations, we show that our schemes can achieve better throughput, less interference to PUs and better load balancing among the channels. In fact, most of the schemes in the literature do not even consider the fact that it is impossible to sense all the channels and the best subset of channels should be selected and made use of.

3.1 System Overview

In the proposed network setting in our scheme, we have a centralized time slotted *Primary Network* (PN) and a decentralized *Secondary Network* (SN) which opportunistically utilizes the PN resources. In this secondary network a REM exists which has the ability to collect the PN channel occupancy details. REM is assumed to possess records of past channel usage information of N channels which are collected from the SUs and a dedicated set of sensors. It is possible for the REM to use an unassigned UHF or VHF channel to collect the channel usage information from the dedicated sensors and the SUs. Since this sensing produce a very small constant bit rate traffic, it is possible to use a dedicated CDMA code in each one of the sensors. Each one of these sensors send a bit stream containing information on whether a PU signal was present or not at the end of each time slot. The economic perspective of the collection of information and the dissemination of the PST models are not the focus of this research. This same channel can be used to send PST models to the SUs.

Further details on the economic perspective can be found in [43] and [41]. The REM having past information on the channel states of N channels, creates PST models of the channel usage and distributes them periodically to the registered SUs as explained above. Here, we adopt the convention of denoting the *busy* period of the PU channel with 1 and the *idle* period with 0. The SUs are assumed to be perfectly synchronized among themselves and with the REM. A diagram of the system topology is shown in Figure 3.1.1.

The SUs were assumed to be equipped with a widely tunable antenna, a sensing unit and a radio unit capable of sensing all the N primary channels and transmitting on the same channels. Furthermore, we assumed that after every Q time slots new models for the channel occupancy of PUs are calculated and distributed by the REM. The SUs and the dedicated sensors were assumed to be capable of differentiating an SU signal from a

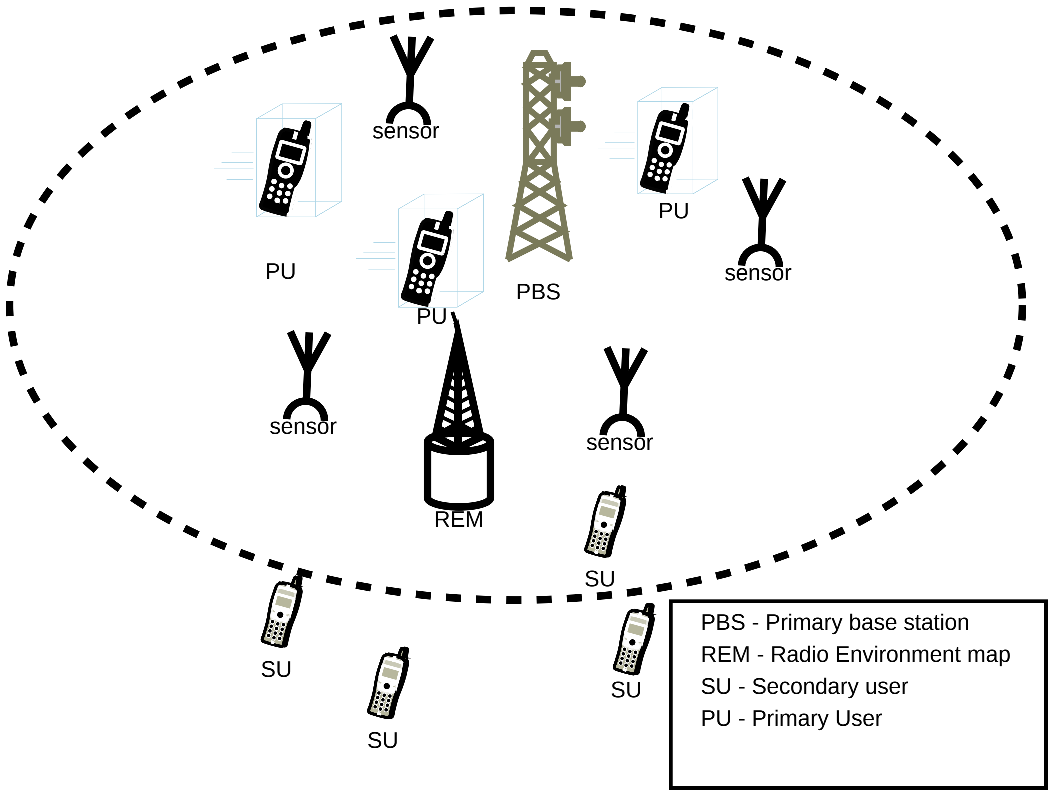


Figure 3.1.1: Overlay network with centralized PN

PU signal. This can be achieved with the use of a universal quiet period for the SUs. In this universal quiet period all the SUs stay silent and sense the channel for the presence of the PU signals. In this scheme we did not assume the existence of a separate common control channel. Rather, we fixed a hopping pattern which was common to all the SUs. This hopping pattern was based on the stationary probability of the channels being idle. Further information on this topic can be found in Section 3.4. The probability distribution learning scheme of the channel states is explained in the next section.

3.2 Channel Modeling and Prediction

Before delving into the details of the learning and prediction schemes, first let us have a look at what is a Variable order Markov model. As we all know, the most commonly used Markov chain in the literature is the one step Markov chain. Now, suppose we let the states of the Markov chain be labeled with a string containing last n observations in the order of occurrence. Then, even though the Markov chain only does one step prediction, that prediction depends on the last n observations, which is called the order of the Markov chain. For example, let '1' and '0' represent the states *busy* and *idle* of a particular channel. A fixed order Markov model of order n has strings of channel observations of the past n time slots as its state labels. Thus, there are 2^n number of states in this particular case. Since the number of states grow exponentially with n, only Markov chains with low order can be practically used [82]. In variable order Markov models, as opposed to fixed order Markov models, we only consider the states which are both highly abundant in the training sequence and have a significant impact on the next observation. As mentioned above the states of the Markov chain are strings of '1's and '0's. Abundance in this context means the number of times a particular string repeats in the training sequence. Therefore, the number of states in variable order Markov models are usually less than those of fixed order Markov models of the same order [83].

If we model a system having a binary state space at a given time point using a variable order Markov model of order $D = 2$, its states are labeled with binary strings of length $l \leq 2$ (strings are binary since the observation space is binary). In this case, the set of states in variable order Markov model is a subset S of $\Lambda = \{0, 1, 00, 10, 01, 11\}$, chosen by an algorithm run on a past observation sequence. This subset S should satisfy a condition that no sequence in it, is a suffix of another sequence in S. For example 1 and 01 both cannot be in set S but 1 and 10 both can be in set S because 1 is not a suffix of 10. A suffix s' of a string s is defined as $s' \in \{s_i s_{i+1} ... s_l | 1 \leq i \leq l\} \cup \{e\}$ where $s = \{s_1 s_2 ... s_l\}$ and e is the null string or empty string [82]. In our case, the observation space is binary. Therefore, $s_i \in \{1, 0\}$ $\forall i \in \{1, 2, ..., l\}$. The other requirement about the state space S is that, if the probability of having $\sigma \in \{1, 0\}$ after s ($Probability(\sigma|s)$) is greater than zero, there should be a unique string $\hat{s} \in S$ which is a suffix of $s\sigma$ for every string $s \in S$, where $s\sigma$ is the concatenation of σ at the end of the string s [82]. As mentioned earlier, to find out the states that matter in the variable order Markov model we use an algorithm called the Probabilistic Suffix Tree (PST) algorithm, which was introduced in [82]. This algorithm builds a tree, which one can use to build a variable order Markov model. To use this algorithm we need a training sequence of sufficient length, which represents channel behavior under normal circumstances. In our case, it would be the channel sensing data gathered by REM, which are binary sequences. We also assumed that REM calculates the mean length of *idle* time of the N channels. The algorithm for the discovery of a *Suffix Set* S, and building the PST, given a training sequence of length m, was explained in the papers [46, 82, 83]. A brief explanation of the algorithm is presented in Appendix A for the convenience of the reader. In this thesis, the symbol appearing at the right hand side end of any observation string is assumed to be the most recent one.

3.2.1 Prediction Using the Probabilistic Suffix Tree Algorithm

For a given PST $\bar{T}$ with maximum Markov order D and the channel observations for the previous k ($0 \leq k \leq D$) consecutive time slots, we traverse the tree starting from node e which represents the empty string. An example tree calculated according to the parameters given in Table 3.1 is shown in Figure 3.2.1. In this parameter list, the first 5 entries are the parameters of the PST algorithm, while σ_1^{10} is an example training sequence of length 10. The nodes of Figure 3.2.1 denotes the conditioning strings selected and the values given in parentheses, $\{p_{idle}, p_{busy}\}$, are the probabilities of occurrence of an idle time slot and a busy time slot after the respective conditioning strings. Then, to find the probability of occurrence of an idle time slot or a busy time slot for a given observation string containing the sensing results of the past consecutive k time slots, we traverse the tree starting from node e. This traversal is done according to the past k channel observations starting with the most recent channel observation and going backwards in time in the past k channel states. Traversing the tree, we reach a leaf node either using all the k channel state observations or partially using them. When this leaf node is reached, we use the probability distribution of the occurrence of an *idle* time slot or a *busy* time slot for that particular node as the probability of occurrence of the next channel state. For example, let us assume the observation string to be '110' and the channel usage model is the one in Figure 3.2.1. Since the most recent observation is 0, we traverse the left branch from e and reach node 0. Then, since the previous observation is 1, we traverse the right hand side branch and reach node 10 which is a leaf node. Therefore, the probability of occurrence of an *idle* time slot after observation string '110' is 0.9994.

In the next section we discuss the MAC protocol and the CCC design used.

Description	Value
P_{min}	0.006
α	0
r	1.05
D	3
γ	0.0006
σ_1^{10}	1110010000

Table 3.1: Threshold values and training sequence used in PST algorithm to obtain the example tree

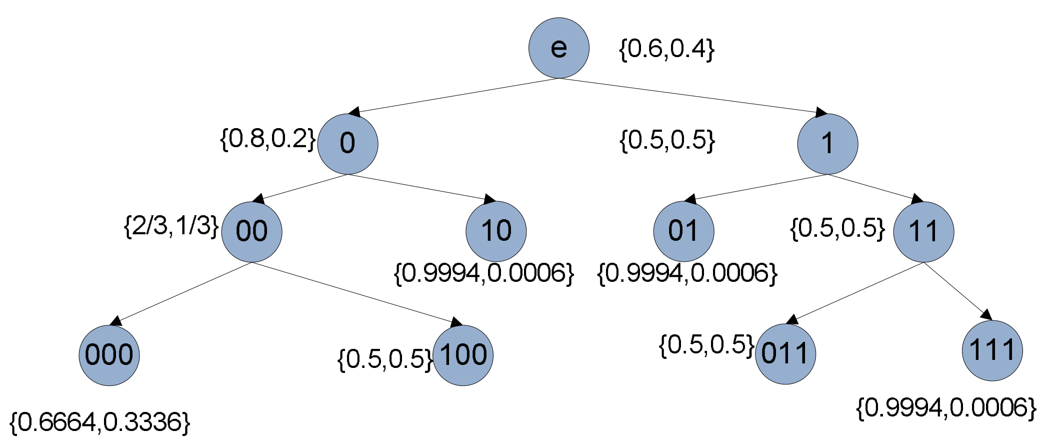


Figure 3.2.1: Example Probabilistic suffix tree

3.3 Proactive Spectrum Access

In the decentralized reactive spectrum access schemes discussed in [44] and [45], SUs using rendezvous channel access schemes search for idle channels in a predetermined or random order. So, after they have selected the channel scan order, they listen to them to identify the state of the channel. If the channel is *idle* the SU used it and if not the search continued until the SU finds a channel which is both idle and the transmitter channel selection coincides with the receiver's. In order to reduce the interruptions to the primary and secondary users, we have to arrange the channels in the ascending order of activity, so that the SUs select the channel with the highest expected availability first, saving them from interfering with the PU and high variability in the idle channel search delay. The channel ordering schemes for the exchange of control information and data are explained in the next subsections.

3.3.1 Channel Ordering Scheme

In the proposed scheme, the communication between two nodes happen in two stages. The first stage is the initial handshake and the second stage is the access stage. In the first stage the transmitter arranges the channels in the descending order of the probabilities of channels being idle. The PSTs of the channels were assumed to be transmitted to the SUs by the REM as mentioned earlier. The stationary idle probability can be found using the PST by finding the probability of the channel being *idle* conditioned on the null string. The parameters of arranging the channels is known to each one of the SUs and that information is identical in each SU. Therefore, switching channels in that order enable an SU pair to communicate whenever the transmitter acquires the channel and, the receiver is tuned to the same channel and is not already communicating with another SU. Further information on the channel acquisition is explained in Section 3.4.

After the completion of this initial hand shake, we assume the two SUs sense and exchange the channel occupancy information of m channels in the current slot. The channel occupancy information at the transmitter is piggybacked on the Data message and the receiver's information is piggybacked on the ACK message. Since we exchange only the binary decisions on the occupancy of the m channels, the communication burden is assumed to be negligible. After the information is exchanged between the transmitter and the receiver, the common information is fused using the OR decision fusion rule. This fused data are used to determine the probabilities of the channels being idle in the next time slot. We arranged these sensed m channels using data of both the PU channel usage and the SU channel usage. More information on this topic is presented in Section 3.5. The number of channels sensed m was upper bounded by the number of channel switches possible in one slot. We fixed the time slot length of the SUs such that it is equal to the minimum of the maximum allowed time between two consecutive channel sensing operations for all N channels. This maximum allowed time between two consecutive sense operations is assumed to be governed by regulations to protect the PUs. Therefore, when the PU communication is slotted, the SU slot length is equal to the minimum of the N PU slot lengths, where there are N PUs in the network.

3.4 MAC Protocol

As discussed in Chapter 2, most MAC protocols in the literature did not take into account the channel usage characteristics of the PUs, with [52, 84] being exceptions. But in those schemes too, the effect of SU access was not taken into account. Therefore, if all the SUs tried to access the optimum channel, the packet collisions in the networks reduce the throughput. In this section, we discuss how the PU channel usage statistics and the SU channel usage statistics could be used towards increasing the throughput of SUs. First, we explain how the PU channel statistics can be used in the CCC sequence selection process. Then, we explain how both the PU and SU statistics can be used in the data transmission process.

3.4.1 Proactive Common Control Channel

Most SH CCC schemes in the literature searched for idle channels in a predetermined or random order. So each SU selected channels based on that channel scan order and listened to them to identify the state of the channel. If the channel was idle the SU used it, and if not the search continued till it finds one and the receiver channel selection coincided with it. In the scheme presented, to reduce the interruptions to the PUs we arranged the channels in the ascending order of activity. Thus, the SUs selected the channel with the highest expected availability more often, saving them from interfering with the PU and high variability in the idle channel search delay.

Let p_i denote the stationary probability of PU channel i being idle, where $i \in \{1, 2, \dots, N\}$. This probability is given by the distribution of channel states at node e of the PST. This node e represents the empty string in the PST. Then, Algorithm 1 is used to generate the channel sequence ϕ_j^t, where ϕ_j^t is the channel selected by SU j at time-slot t.

Algorithm 1 Control channel selection algorithm

1. Initialize $\phi_j^t = \emptyset$
2. $u \Leftarrow rand()$
3. **for** $k = \{1, 2, 3, \dots, N\}$ **do**
4.   **if** $u \leq \frac{\sum_{l=1}^{k} p_i}{\sum_{i=1}^{N} p_i}$ **then**
5. $\phi_j^t = k$
6.   break;
7.   **end if**
8. **end for**

In the above scheme the SUs selected the channels with less PU activity more often. Furthermore, due to the randomness in the selection process, the number of packet collisions was low. The downside to this channel selection method is, not having a guaranteed rendezvous time as the schemes [44] and [85].

Analytic Formulation of the Average Time to Rendezvous

The time to rendezvous is an important performance metric because it decides the number of attempts an SU has to make to successfully complete a handshake with the receiver. This metric dictates the packet delays. Therefore, time to rendezvous should be as small as possible. In order to derive a formula for the average time to rendezvous for a transmitting SU, first, the probability of successfully contacting the receiver should be calculated. The method of calculation is explained below.

First, the following assumptions were made: the contention window size is CW, all back-off time values are equi-probable with probability $\frac{1}{CW+1}$ and the total number of SUs is M. The vulnerable time is given by t_v. This is the minimum time gap between the smallest back-off time window and the second smallest back-off time window of contending neighbors on the same channel to avoid a packet collision. In other words, this is the average time duration it takes for nodes in the CSMA/CA network to identify the channel is busy, after the first node transmits. This time is equal to the sum of the transmission time duration of the RTS packet and the propagation time, in the unit of sub-time slots (smallest synchronous time unit) [86]. The false alarm probability is given by symbol p_{fa} and the probability of selecting channel $k(\leq N)$ is given by p_k^s. According to the CCC selection algorithm above $p_k^s = \frac{p_k}{\sum_{i=1}^N p_i}$. The probability of channel access by an SU is given by α and the cumulative distribution of the random back-off time slots generated is given by $F(\cdot)$. Using these parameters the formula for the probability of successful handshake, θ_s, was formulated as in Equation (3.4.1).

$$\begin{aligned}\theta_s &= \sum_{i=1}^{N} p_i p_i^s \left[\sum_{r=0}^{M-1} (1-p_{fa}) \binom{M-1}{r} (\alpha p_i^s)^r (1-\alpha p_i^s)^{M-r-1} \right. \\ &\quad \left. \times \sum_{\tau=0}^{CW} (1-F(\tau+t_v))^r \frac{1}{CW+1} \right] p_i^s (1-\alpha) \,. \end{aligned} \tag{3.4.1}$$

The steps of the derivation of this equation are given below.

Proof. First, let us find the conditions that should be met in order for an SU to succeed in the handshake. A given SU j can succeed in the handshake if all the conditions below are satisfied:

- SU j has a packet to transmit, which is given by probability α.
- The SU j selects channel i, which is given by the probability p_i^s.
- Channel i is idle, given by probability p_i.
- Channel i is sensed as idle by SU j, given by probability $(1 - p_{fa})$.

- SU j wins the contention.
- The receiver has selected channel i and is not transmitting, given by probability $p_i^s \times (1-\alpha)$.

For the SU j to win the contention, the back-off duration of it should be less than the minimum of all the other SUs trying to transmit in the same channel by at least the vulnerable time t_v. In this paper, we assumed all the SUs can hear each other. If the cumulative distribution function of the back-off duration is given by $F(\cdot)$, and SU j's back off duration is τ, the probability $R(\tau + t_v)_r$, that r number of transmitting SUs back-off times are more than $\tau + t_v$ is given by:

$$R(\tau + t_v)_r = (1 - F(\tau + t_v))^r \ . \tag{3.4.2}$$

Then the distribution of r SUs trying to access channel i can be derived using the binomial distribution with probability of success $\alpha \times p_i^s$ and total number of other SUs $M-1$ as in Equation (3.4.3).

$$B(M-1, \alpha p_i^s) = \binom{M-1}{r} (\alpha p_i^s)^r (1 - \alpha p_i^s)^{M-r-1} \ . \tag{3.4.3}$$

Equation (3.4.2) shows the probability of an SU winning the contention, which is conditioned on: the back-off time τ of the SU, the number of other SUs, r, trying to access a given channel and the given SU trying to transmit on the given channel. Since the back-off time is uniformly distributed in $[0, CW]$, we took the expectation of $R(\tau+t_v)_r$ with respect to the back off duration and the number of other users trying to access channel i. The result is the probability of an SU winning the contention on channel i, which is stated in Equation (3.4.4).

$$\begin{aligned}\zeta_s(i) &= \sum_{r=0}^{M-1} \binom{M-1}{r} \left(\alpha p_i^s\right)^r \left(1-\alpha p_i^s\right)^{M-r-1} \\ &\quad \times \sum_{\tau=0}^{CW} \left(1-F(\tau+t_v)\right)^r \frac{1}{CW+1} \end{aligned} \tag{3.4.4}$$

Then, multiplying $\zeta_s(i)$ by the probability of the receiver being tuned into channel i, αp_i^s, we get the probability of successful handshake in channel i. Finally, we took the expectation of it w. r. t. all the channels, where the probability of selecting a given channel was $p_i\left(1-p_{fa}\right)p_i^s$. This gave us the result shown in Equation (3.4.1).

□

Then, the probability distribution of the number of time slots to achieve rendezvous $TR(q=l)$ was calculated as shown in Equation (3.4.5).

$$TR(q=l) = \left(1-\theta_s\right)^{l-1} \theta_s \, . \tag{3.4.5}$$

Channel Load in the SH Control Channel

Since this scheme used a stochastically selected control channel, only the average channel load can be calculated. The probability of r users out of M SUs accessing channel i can be given by $B(M, \alpha p_i^s)$ which was defined in Equation (3.4.3). Then, using $B(M, \alpha p_i^s)$, an equation for the maximum average channel load was formulated as shown in Equation (3.4.6).

$$L\left(M, \underline{\mathbf{p}}^s, \alpha\right) = \max_{i=\{1,2,\cdots,N\}} \sum_{r=1}^{M} r \cdot B(M, \alpha p_i^s) \, . \tag{3.4.6}$$

Degree of Rendezvous

This metric gives the number of minimum overlaps between any two channel sequences in a SH based CCC. In a stochastic channel selection scheme, although a number for the degree of rendezvous cannot be given, the probability for two sequences of length Q to have a minimum of k overlaps can be calculated. Equation (3.4.7) gives an expression for this probabilistic degree of rendezvous.

$$DR(Q,k,\underline{\mathbf{p}}^s) = \sum_{i=1}^{N} p_i^s \sum_{r=k}^{Q} \binom{Q}{r} (p_i^s)^{2r} \left(1-(p_i^s)^2\right)^{Q-r} \ . \qquad (3.4.7)$$

3.5 Channel Set Selection Scheme for the Data Phase

In this scheme, we propose two methods to select channels for the data phase. The first method has a very low computational complexity and the channel selection depends only on the past channel state information. The second scheme is based on the reward each channel generates for Q time slots in the future, which has a higher computational complexity.

In proactive channel access schemes, the SUs have to keep track of multiple channels in order for them to switch their transmissions into a different channel either when the PU reappears on the current channel or when there are better channels available. Since the sensing operation uses SU resources such as transmission time and power, we limit the number of channels to keep track of to be equal to the maximum number of channel switches m that can be made during a time slot. Then, we select m channels to sense based on the two methods described below.

3.5.1 Method-I : Channel Set Selection Based on Past Data

In predicting the status of a channel using the PST, we need to sense the channels. To make the CR less expensive we use a single half-duplex transceiver and do energy detection based narrow band sensing. Therefore, in our scheme only sequential sensing is possible and there is an upper bound m, to the number of channels an SU could sense in a given time slot. This upper bound is the time needed to fail in transmitting $m-1$ times and trying transmitting for the m^{th} time, so that there is time for the contention window, RTS and CTS messages , data and the ACK. The selection of a set of m channels, which are both utilized less by the PU and the other SUs, is necessary to achieve a better throughput. To this end, we propose a channel set selection scheme which choose highly available channels for a given SU j, with high probability. To find this channel set, we have to calculate the probability of winning the contention on a given idle channel i.

Calculating the Probability of Winning the Contention

This is a subjective usage indicator on an SU j of interest. To calculate this parameter each SU kept track of the instances it tried to access channel i, ϵ_{ij}^{tot} and the instances it acquired channel i, ϵ_{ij}^{acq}. Then for the channels the SU have been accessing, we calculated c_{ij} using the ratio between the two, $\frac{\epsilon_{ij}^{acq}}{\epsilon_{ij}^{tot}}$. For the channels the SU have not tried to access, c_{ij} is calculated based on the probability a channel is successfully accessed by an SU, given that each SU chose channels according to Algorithm 1. Therefore, channel i is assumed to be accessed with probability $\frac{\alpha p_i}{\sum_{i=1}^{N} p_i}$. Thus, for the unused channel i and SU j the parameter c_{ij} is calculated as given in Equation (3.5.1).

$$
\begin{aligned}
c_{ij} \quad = \quad & \sum_{r=0}^{M-1} (1 - p_{fa}) \binom{M-1}{r} \left(\frac{\alpha p_i}{\sum_{i=1}^{N} p_i} \right)^r \left(1 - \frac{\alpha p_i}{\sum_{i=1}^{N} p_i} \right)^{M-r-1} \\
& \times \sum_{\tau=0}^{CW} (1 - F(\tau + t_v))^r \frac{1}{CW+1} \,. \qquad (3.5.1)
\end{aligned}
$$

Algorithm for Channel Set Selection

Using the above calculated parameter c_{ij} and the stationary probability of channel i being idle, p_i, we select the channel set using Algorithm 2. Here, $\mathcal{S}_j$ is the selected channel set and $\bar{\mathcal{S}}_j$ is the set of channels which are not selected.

Algorithm 2 Channel set selection algorithm

Initialize $\mathcal{S}_j = \emptyset$
while $|\mathcal{S}_j| < m - 1$ **do**
  $\bar{\mathcal{S}}_j = \{1, 2, 3, \ldots, N\}/\mathcal{S}_j$
  $k = \underset{i \in \bar{\mathcal{S}}_j}{\arg\max}\{p_i c_{ij}\}$
  $\mathcal{S}_j = \mathcal{S}_j \cup \{k\}$
end while

3.5.2 Method-II : Channel Set Selection Based on Phase-Type Distribution

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In this section, we develop a scheme to select a set of channels for data transmission, taking into account the potential net reward each channel generates. The method of calculating this reward is explained below. In the control phase, SUs sense the channels in the order discussed earlier, and find them either occupied by the PU or idle. Therefore, SUs have the knowledge of the state of a particular channel they have already sensed prior to l time slots. Furthermore, they know the stationary distribution of the channel occupation of the PUs on channels $i \in \{1, 2, 3, ..., N\}$, π_i, and the transition probability matrix T_i^{PU}, which governs the changes in the channel state, for all the channels including the ones they did not sense. This T_i^{PU} can be generated with the use of PSTs since the PST has information on the conditional probability of the occurrence of either a *busy* or an *idle* state conditioned on the previous state. The information on the channel state prior to t time slots or the stationary distribution together with the channel state transition matrix, T_i^{PU}, can be used to calculate the probability of the channel being idle in the current time slot. Furthermore, if each SU j knows the probability that a given SU wins the contention on channel i when the channel was idle, c_{ij}, we can create a Markov chain for the state transitions for a given channel, assuming each SU uses a single channel from start to the finish of its transmission.

We assumed here that, the channel transition probability due to PU activities can be explained sufficiently with a 1^{st} order Markov chain, where p^i_{00} represents the probability of channel transition from idle state to idle state, p^i_{01} the probability of channel transition from idle state to busy state, p^i_{10} the probability of channel transition from busy state to idle state and p^i_{11} the probability of channel transition from busy state to busy state. The stationary probability distribution of the channel states were assumed to be $\pi_i = [p^i_1, p^i_0]$. This assumption is not far from reality, as it was observed in [87] that the largest improvement in predictability occurred when going from 0^{th} order to 1^{st} order in the majority of channels they surveyed.

$$T_i^{PU} = \begin{array}{c c} & \begin{array}{cc} \textit{busy} & \textit{idle} \end{array} \\ \begin{array}{c} \textit{busy} \\ \textit{idle} \end{array} & \begin{pmatrix} p^i_{11} & p^i_{10} \\ p^i_{01} & p^i_{00} \end{pmatrix} \end{array}. \tag{3.5.2}$$

If an SU generates a packet in a given time slot with probability α, wins the contention on channel i in the time slot with probability c_i, and p_{fa} is the false alarm probability, we can come up with the transition probability matrix for channel i as shown in Equation (3.5.3) on page 61, where the mathematical formulae for the entries in this matrix are explained in Equation (3.5.4) on page 61. Here, we have dropped the subscript j in c_{ij} since we are focusing on a given SU here onwards. This transition matrix represents the probabilities of the packet fluctuations in the SU queue on channel i for a given transmission. We assumed that the transmission session ends once the queue gets empty. The states of the Markov chain are labeled as (n, s), where n is the number of packets in the queue and s is the actual state of the PU channel. We further assume that the data queue of the SU is finite having capacity of K, when developing this transition matrix.

$$
T_i^{SU} = \begin{array}{c} \\ 0 \\ (1,1) \\ (1,0) \\ (2,1) \\ (2,0) \\ \vdots \\ (K,1) \\ (K,0) \end{array}
\begin{array}{c}
\begin{array}{cccccccccc} 0 & (1,1) & (1,0) & (2,1) & (2,0) & (3,1) & (3,0) & \cdots & (K,1) & (K,0) \end{array} \\
\begin{pmatrix}
1 & 0 & 0 & 0 & 0 & 0 & 0 & \cdots & 0 & 0 \\
0 & a_1^1 & a_2^1 & a_1^2 & a_2^2 & 0 & 0 & \cdots & 0 & 0 \\
d_1 & a_3^1 & a_4^1 & a_3^2 & a_4^2 & 0 & 0 & \cdots & 0 & 0 \\
0 & 0 & 0 & a_1^1 & a_2^1 & a_1^2 & a_2^2 & \cdots & 0 & 0 \\
0 & a_1^0 & a_2^0 & a_3^1 & a_4^1 & a_3^2 & a_4^2 & \cdots & 0 & 0 \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \cdots & \vdots & \vdots \\
0 & 0 & 0 & 0 & 0 & 0 & 0 & \cdots & a_1^1 + a_1^2 & a_2^1 + a_2^2 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 & \cdots & a_3^1 + a_3^2 & a_4^1 + a_4^2
\end{pmatrix}
\end{array},
\tag{3.5.3}
$$

$$
\begin{aligned}
d_1 &= (1-\alpha)c_i(p_{00}^i + p_{01}^i)(1-p_{fa}) \, , \\
a_1^1 &= (1-\alpha)p_{11}^i \, , \\
a_2^1 &= (1-\alpha)p_{10}^i \, , \\
a_3^1 &= \alpha c_i(1-p_{fa})p_{01}^i + (1-\alpha)\left((1-c_i)(1-p_{fa}) + p_f a\right)p_{01}^i \, , \\
a_4^1 &= \alpha c_i(1-p_{fa})p_{00}^i + (1-\alpha)\left((1-c_i)(1-p_{fa}) + p_f a\right)p_{00}^i \, , \\
a_1^2 &= \alpha p_{11}^i \, , \\
a_2^2 &= \alpha p_{10}^i \, , \\
a_3^2 &= \alpha\left((1-c_i)(1-p_{fa}) + p_{fa}\right)p_{01}^i \, , \\
a_4^2 &= \alpha\left((1-c_i)(1-p_{fa}) + p_{fa}\right)p_{00}^i \, , \\
a_1^0 &= (1-\alpha)c_i(1-p_{fa})p_{01}^i \, , \\
a_2^0 &= (1-\alpha)c_i(1-p_{fa})p_{00}^i \, .
\end{aligned}
\tag{3.5.4}
$$

This Markov chain represents an absorbing Markov chain, where the State 0 is the absorbing state. Since the system starts in either state $(1,1)$ or $(1,0)$, we can calculate the initial vector τ^i for channel i, either based on the observed state for channel i or based on the stationary state probability for channel i. Given that there is an absorbing state and several transient states in the transition matrix in Equation (3.5.3), we can partition it and create a Phase-Type distribution (τ^i, T^i) as explained in the next section.

Developing the Ph-Type Distribution to Calculate the Reward of Using a Channel

The transition matrix given above can be converted to a Phase type distribution with the parameters (τ^i, T^i). The transient part T^i is represented in Equation (3.5.5), where the block matrices, A_0, A_1 and A_2 are given in Equation (3.5.6).

$$T^i = \begin{bmatrix} A_1 & A_2 & 0 & 0 & \cdots & 0 \\ A_0 & A_1 & A_2 & 0 & \cdots & 0 \\ 0 & A_0 & A_1 & A_2 & \cdots & 0 \\ \vdots & \vdots & \vdots & \vdots & \cdots & \vdots \\ 0 & 0 & 0 & 0 & \cdots & A_1 + A_2 \end{bmatrix}, \tag{3.5.5}$$

$$\begin{aligned} A_0 &= \begin{bmatrix} 0 & 0 \\ a_1^0 & a_2^0 \end{bmatrix}, \\ A_1 &= \begin{bmatrix} a_1^1 & a_2^1 \\ a_3^1 & a_4^1 \end{bmatrix}, \\ A_2 &= \begin{bmatrix} a_1^2 & a_2^2 \\ a_3^2 & a_4^2 \end{bmatrix}. \end{aligned} \tag{3.5.6}$$

The absorption vector in the Ph-Type distribution T_0^i can be represented as in Equation (3.5.7).

$$T_0^i = \begin{bmatrix} 0 \\ d_1 \\ 0 \\ \vdots \\ 0 \end{bmatrix} . \tag{3.5.7}$$

If channel i was sensed to be busy at time slot κ, its probability state vector at the time of sensing can be given by τ_B^i. If it was sensed to be idle, the state vector can be given by τ_I^i. The vectors τ_B^i and τ_I^i are presented in equations (3.5.8) and (3.5.9) respectively, where p_m represents the missed detection probability.

$$\tau_B^i = \begin{bmatrix} (1-p_m)p_1^i \\ 1-(1-p_m)p_1^i \end{bmatrix} , \tag{3.5.8}$$

$$\tau_I^i = \begin{bmatrix} 1-(1-p_{fa})p_0^i \\ (1-p_{fa})p_0^i \end{bmatrix} . \tag{3.5.9}$$

Then, we can convert this probability vector to the one at time $n(>\kappa)$ using Equation (3.5.10), where $\theta = \{I, B\}$ and T stands for the transpose operation.

$$\tau_n^i = \left(\tau_\theta^i\right)^T \begin{bmatrix} p_{11}^i & p_{10}^i \\ p_{01}^i & p_{00}^i \end{bmatrix}^{n-\kappa} . \tag{3.5.10}$$

For the channels which were not sensed, the state probability vector is calculated as in Equation (3.5.11).

$$\tau_n^i = \begin{bmatrix} p_1^i \\ p_0^i \end{bmatrix}^T . \tag{3.5.11}$$

Then the initial vector for the Ph-type distribution τ^i can be presented as in Equation (3.5.12).

$$\tau^i = \begin{bmatrix} \tau_n^i & 0 & \dots & 0 \end{bmatrix} . \tag{3.5.12}$$

Calculating the Reward Using the Ph-Type Distribution

Given the Ph-type distribution with parameters (τ^i, T^i), we used the method used in [88] to calculate the benefit of using channel i. This calculation was based on marked Markov chains. Then we picked the channels which have the highest net reward. This method of calculating the net reward is explained below for completeness. In this method, we calculate the probability of a sequence of events happening which includes a specified event of interest happening. This is done using a dummy variable as a marker for the event of interest. In the scenario we investigated in this thesis, we are interested in whether an SU was successful in transmission or if it failed to transmit in a given time slot. If it was able to transmit in a given time slot, it earns a reward, otherwise it incurs a penalty. Thus, the channels with the highest net rewards are the best channels to use. For the purpose of identifying the instances a packet gets transmitted, we introduced a dummy variable z and changed the transient matrix T^i to $T_i^D(z)$. The only difference in $T_i^D(z)$ is that, A_0 and A_1 in T^i are replaced by $A_0^D(z)$ and $A_1^D(z)$ in $T_i^D(z)$ respectively. The matrices $T_i^D(z)$, $A_0^D(z)$ and $A_1^D(z)$ are shown in equations (3.5.13) and (3.5.14) respectively.

$$T_i^D(z) = \begin{bmatrix} A_1^D(z) & A_2 & 0 & 0 & \cdots & 0 \\ A_0^D(z) & A_1^D(z) & A_2 & 0 & \cdots & 0 \\ 0 & A_0^D(z) & A_1^D(z) & A_2 & \cdots & 0 \\ \vdots & \vdots & \vdots & \vdots & \cdots & \vdots \\ 0 & 0 & 0 & 0 & \cdots & A_1 + A_2 \end{bmatrix} , \tag{3.5.13}$$

$$A_0^D(z) = \begin{bmatrix} 0 & 0 \\ a_1^0 \cdot z & a_2^0 \cdot z \end{bmatrix}, \tag{3.5.14}$$

$$A_1^D(z) = \begin{bmatrix} 0 & 0 \\ a_1^0 \cdot z & a_2^0 \cdot z \end{bmatrix}. \tag{3.5.15}$$

For the case of failing to transmit, we changed the matrix T^i to $T_i^U(z)$. The only difference in $T_i^U(z)$ was that A_1 and A_2 in T^i getting replaced by $A_1^U(z)$ and $A_2^U(z)$ in $T_i^U(z)$ respectively. The matrices $T_i^U(z)$, $A_1^U(z)$ and $A_2^U(z)$ are stated in equations (3.5.16) and (3.5.17) respectively. The entries in Equation (3.5.17) are defined in Equation (3.5.18). In the case of failing to transmit, we do not consider the effect of the probability of mis-detection, assuming this probability is the same for all the channels. There is no gain in identifying when SUs try to transmit when the channel was busy, because the transmission fails with certainty. If the probability of mis-detection or the probability of false alarm is different for each channel, we can use the same method to penalize the channel. Therefore, here we only penalize the events of not being able to transmit due to the PU or another SU being active. Then, we calculate the reward of transmission using the same method and calculated the net reward.

$$T_i^U(z) = \begin{bmatrix} A_1 & A_2 & 0 & \cdots & 0 \\ A_0 & A_1^U(z) & A_2^U(z) & \cdots & 0 \\ 0 & A_0 & A_1^U(z) & \cdots & 0 \\ \vdots & \vdots & \vdots & \cdots & \vdots \\ 0 & 0 & 0 & \cdots & A_1^U(z) + A_2^U(z) \end{bmatrix}, \tag{3.5.16}$$

$$
\begin{aligned}
A_1^U(z) &= \begin{bmatrix} a_1^1 \cdot z & a_2^1 \cdot z \\ a_3^1(z) & a_4^1(z) \end{bmatrix}, \\
A_2^U(z) &= \begin{bmatrix} a_1^2 \cdot z & a_2^2 \cdot z \\ a_3^2 \cdot z & a_4^2 \cdot z \end{bmatrix},
\end{aligned} \tag{3.5.17}
$$

$$
\begin{aligned}
a_3^1(z) &= \alpha c_i(1-p_{fa})p_{01}^i + (1-\alpha)\left((1-c_i)(1-p_{fa}) \cdot z + p_f a\right) p_{01}^i \,, \\
a_4^1(z) &= \alpha c_i(1-p_{fa})p_{00}^i + (1-\alpha)\left((1-c_i)(1-p_{fa}) \cdot z + p_f a\right) p_{00}^i \,.
\end{aligned} \tag{3.5.18}
$$

After coming up with T_i^D and T_i^U which stand for reward and penalty respectively, we calculate the expected reward and penalty for each channel. If the SU took $\eta\,(\geq 2)$ time slots to finish the transmission, the expected number of time slots the packet transmission was successful, $R^i(\eta)$, is calculated as shown in Equation (3.5.19). Similarly, the expected number of time slots the packets are not transmitted, $P^i(\eta)$, are calculated as shown in Equation (3.5.20).

$$
R^i(\eta) = \frac{\partial}{\partial z}\left[\tau^i \left\{\sum_{l=1}^{\eta-1} (T^i)^{l-1} T_i^D(z) (T^i)^{\eta-l-1}\right\} T_0^i\right], \tag{3.5.19}
$$

$$
P^i(\eta) = \frac{\partial}{\partial z}\left[\tau^i \left\{\sum_{l=1}^{\eta-1} (T^i)^{l-1} T_i^U(z) (T^i)^{\eta-l-1}\right\} T_0^i\right]. \tag{3.5.20}
$$

Given that SU took η slots to finish the transmission, the net reward is calculated taking the difference of $R^i(\eta)$ and $P^i(\eta)$. Then, taking the sum of the difference between the two for values of η from 1 to Q, we calculated the net reward $N^i(Q)$ of using channel i, given that the transmission time is less than Q, as shown in Equation (3.5.21). This Q is the time duration within which the PU channel usage model is valid.

$$N^i(Q) = \sum_{\eta=1}^{Q} R^i(\eta) - P^i(\eta)\ . \tag{3.5.21}$$

Then we sorted the channels in the descending order of $N^i(Q)$, and found the best m channels to keep track of, to perform proactive switching. This m is equal to the number of channel switches that can be performed within a time slot by the SU.

Complexity of the Channel Selection Method

In this channel selection method multiple number of matrix multiplications are carried out along with partial differentiation. This partial differentiation does not add more complexity as only the matrices $T_i^D(z)$ in the reward function and $T_i^U(z)$ in the penalty function have variable z in them and we can perform the differentiation prior to multiplication. Differentiation in this context meant replacing the constant terms with zeros, and terms multiplied by the dummy variable z, by only the coefficient of z. The complexity of multiplying two $n \times n$ matrices is given by $O(n^3)$. In our case $n = 2 \cdot K + 1$, where K is the buffer size. Then $\sum_{\eta=3}^{Q}(\eta - 3)(\eta - 1)$ number of multiplications should be done if we do not save the results of previous ones to calculate the reward and penalty. So at the worst case, the complexity is a cubic function of both Q and n. But the matrices were sparse.

In the following section we discuss how the communication between two SUs happen.

3.5.3 Communication Between Two SUs

In the scheme presented, the communication between two nodes happen in two stages. First, an initial handshake between SUs is completed and then the SUs communicate data between each other. In the first stage the transmitting SU generate the channels to be used at each time slot t, ϕ_j^t, using Algorithm 1. Then, it generates the channel set $\mathcal{S}_j$ to be used for data transmission, either using Algorithm 2 or method explained in Section 3.5.2. After that, the transmitting SU u, start the handshake phase with receiver v. This procedure is explained with more details in the next subsection.

Handshake Between the Sender and the Receiver

Prior to the handshake with the receiver, in the QP, the transmitter u first senses the channel list $\mathcal{S}_j$, and then senses the control channel ϕ_j^t. Then, in the back-off period, it calculates the statistical availability of the channel set $\mathcal{S}_j \cup \phi_j^t$ at time slot $t+1$ using PST algorithm. This ordered list H_u^{t+1} is piggybacked on the RTS packet. If the intended receiver happened to listen to the same channel (i.e.: $\phi_u^t = \phi_v^t$) and was able to decode the packet, it transmits a CTS packet. Then the data communication is done using the same channel ϕ_j^t till the end of the time slot and the receiver sends an ACK. At this stage the handshake is considered to be successful. If the CTS gets timed out, the procedure is repeated in the next time slot. Figure 3.5.1 shows an example of transmitter slot structure at the handshake phase.

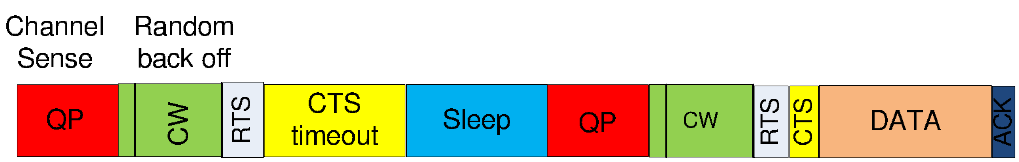

Figure 3.5.1: An example of transmitter slot structure at handshake phase

Data Transmission After the Successful Handshake

If the handshake was successful at time slot t, the receiver v and transmitter u knowing the channels to be used in the data access phase, sense the channel set H_u^{t+1} in the quiet period of time slot $t+1$. This sensing is done starting from the worst statistical availability to the best (i.e. in the reverse order). Therefore, after sensing the best channel the transmitter u generates the back-off time, and if sensed idle it will send the RTS with the sensed channel status data on channel set H_u^{t+1} piggybacked on it. Then, if this RTS was received at the receiver v, it replies with a CTS with sensed channel status data on set H_u^{t+1} piggybacked on it. Then, after the data transmission starts, at the end of the time slot the receiver v sends an ACK if the transmission was successful. If the CTS was not received at the transmitter, the transmitter tries the next channel in sequence H_u^{t+1} after the CTS times out. This goes on until the list H_u^{t+1} gets exhausted within time slot $t+1$. We set this time out period to be equal to the sum of CTS transmission time and propagation delay, and the time to transmit the header of the data part and propagation delay. If the transmitter u was unable to contact the receiver v for an entire time slot then the initial handshake has to be redone, because the channel status at both ends of the current slot are not known to both u and v.

If the RTS and CTS exchange was successful, the sensing outcomes of the channels at both ends are available to both u and v. Then, both u and v fuse the sensing results using OR decision fusion and save this result for each channel i separately in a buffer s_{uv}^{i}, where $i \in \mathcal{S}_j \cup \phi_j^t$. Each user u clears the buffer s_{uv}^{i} when the channel i is not sensed or the communication between each other is over in that time slot. Then this buffer content at time slot $t+1$ is used to predict the channel status at time $t+2$. Then both u and v make their ordered lists H_u^{t+2} and H_v^{t+2}. Both of these lists are the same. Therefore in time slot $t+2$ the communication is possible. This process goes on till the communication between u and v is complete. A diagram showing an example of transmitter slot structure at data transmission phase is shown in Figure 3.5.2.

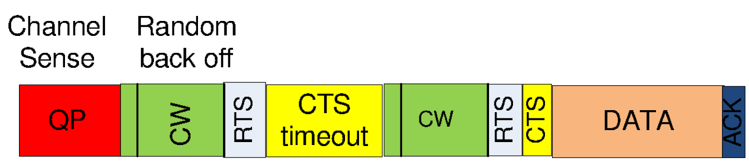

Figure 3.5.2: An example of transmitter slot structure at data phase

3.5.4 Registration of New Users, Synchronization and Model Updates

We assumed that the REM transmits beacons every T time slots on all the idle PU channels. This beacon transmission starts at the QP of the SUs. Therefore, it is easy for the SUs to detect this beacon. This beacon contains a synchronizing bit sequence and encoded channel models. The SUs who have registered with the REM can decode these models. In these beacons the periodic channel hopping pattern of the REM is mentioned. Therefore the SUs newly arriving to the network is able to follow the REM hopping pattern and communicate with it to get registered. A newly arriving SU scans the spectrum until it receives the beacon. The period T was determined as the minimum of the time for 5% of SUs to have a clock drift of $\frac{QP}{2}$ and the inter-arrival time between two new user arrivals to have 95% probability of arrival. Since all the registered SUs get synchronized with the REM from time to time the assumption of the SUs being synchronous with each other holds true.

3.6 Simulation

In this section, we carry out the numerical evaluation of the proposed scheme. In order to create the channel model according to the probabilistic suffix tree scheme, we needed a training sequence. Real data was not available to us. So, we created a set of training data as strings of '1's and '0's. We used a renewal process with *busy* time and an *idle* time generated according to a pseudo self-similar Markovian Arrival process (MAP) to create the training data. The used Markovian Arrival Process (MAP) is capable of capturing the correlation between the different channel states. We developed the MAP as explained in [64, 89] for the simulation purpose. As mentioned above, this MAP exhibited what is known as Pseudo long range dependent self-similar characteristics. Self-similarity means that the burstiness of traffic is the same over different time scales. In other words, if we measure the traffic in a particular link and plot the traffic as a graph of the average amount of packets per 1s, 10s, 100s, 1000s etc. the graph will only be scaled in amplitude, while the shape remains the same. The structure of the Markovian arrival process explained in [64] had burstiness similar to Ethernet traffic. The MAP distribution is defined by a stochastic matrix D as defined in Equation (3.6.1), where D_0 and D_1 are sub-stochastic matrices containing the state transition probabilities to phases within the idle state and busy state respectively and d_{01} (d_{10}) contains the probabilities of transition to a phase in busy (idle) state from a phase in idle (busy) state. We fixed the size of the D matrix to be 6×6. The parameters of the MAP were calculated such that the channel utilization are $0.2, 0.4$ and 0.8, and the mean burst length is 10 slots. The theoretical properties of this MAP

SNR Range	Modulation scheme	Packets transmitted
$-\infty dB$ - $9dB$	BPSK	0
$9dB$ - $12dB$	BPSK	1
$12dB$ - $17dB$	QPSK	2
$17dB$ - ∞dB	8-PSK	3

Table 3.2: Parameters used in the AMC scheme

distribution can be found in [64].

$$D = \begin{bmatrix} D_0 & d_{01} \\ d_{10} & D_1 \end{bmatrix} \tag{3.6.1}$$

Before training the models, we changed the *busy* slots to *idle* with probability P_m and *idle* slots to *busy* with probability P_f in the channel state sequence generated. The false alarm probability P_f was calculated for a given missed detection probability, $P_m = 0.1$, assuming an AWGN channel with mean SNR $5dB$ at the SU. Then we simulated the channel access of 5, 10 and 20 SU pairs using 10 channels, while varying the SU channel access probability values, α, from 0.1 to 0.9. In the simulations, we assumed the distance between any two communicating SU pairs to be uniformly distributed between 10 meters and 50 meters. Then, the channel gain was calculated assuming a path loss exponent of 2.7, a log-normal shadowing loss with a standard deviation of 11.8 and Rayleigh fading. We calculated the signal to noise ratio (SNR) assuming a transmit power of $100mW$ and a noise power of $0.001mW$. Then we assumed an adaptive modulation and coding (AMC) scheme where the details of SNR ranges, modulation scheme used and the number of packets transmitted per sub-slot are listed in Table 3.2. In the simulations, we assumed that the channel gain information is perfectly known at the transmitter.

We used the parameters given in Table 3.3 when training the PST and in the simulation setup. In the graphs presented, our two schemes are labeled *Method-I* and *Method-II*, while the random channel set selection scheme is named *Random*. In the *Random* channel set selection scheme, we used the scheme introduced in [57] to predict the channel status. Since the schemes in the literature that we looked at did not consider the problem of channel set selection, we assumed random channel set selection together with the hidden Markov model (HMM) based scheme in [57] as the current state of the art. The length of the stringof observations used for training was 20000 for both the PST scheme the HMM scheme. These schemes were simulated for 50000 time slots and this entire experiment was repeated 8 times to get a mean and error plot.

Parameter	Value
D	10
P_{min}	0.006
α	0
r	1.05
γ	0.0006
N	10
k	3
Channel 1-4 PU utilization	0.2
Channel 5-7 PU utilization	0.4
Channel 8-10 PU utilization	0.8
Probability of missed detection	0.1

Table 3.3: Simulation Parameters

3.6.1 Initial Handshake

We used the control channel selection scheme introduced in [85] (*RCCH-Sync*) when running simulations for the random channel set selection scheme. Then, we used the control channel selection scheme introduced in this paper when simulating Method-I and Method-II. In figures 3.6.1, 3.6.2 and 3.6.3, we plot the average number of time slots taken by each scheme to achieve rendezvous for 5, 10 and 20 SUs respectively.

One can see in figures 3.6.1, 3.6.2 and 3.6.3 that our scheme performs as equally well as the *RCCH-Sync* scheme. For this comparison we used the scheme developed for synchronized networks in [85]. In that scheme, the synchronization off-set between the transmitter and the receiver could not even have values which are integer multiples of the time-slot length. Our scheme can handle synchronization issues of that form. In [85], authors developed schemes which can withstand time offsets which are integer multiples of the time-slot

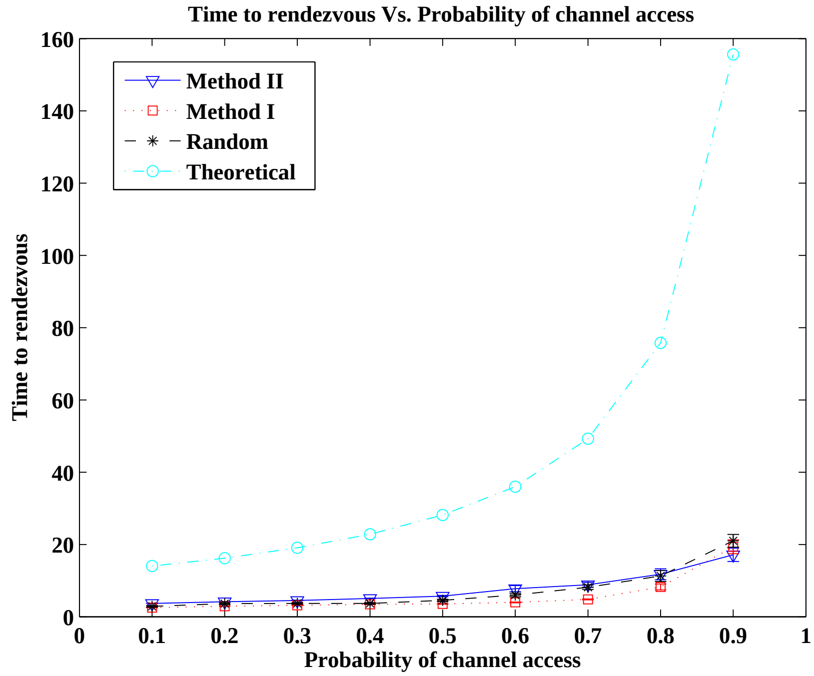

Figure 3.6.1: Average number of control slots Vs. SU channel access probability for 5 SUs

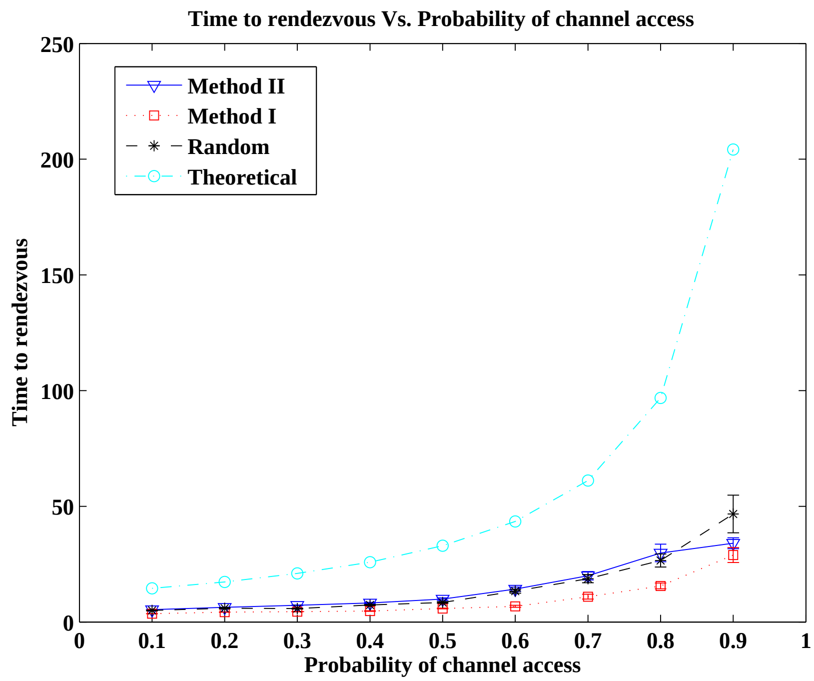

Figure 3.6.2: Average number of control slots Vs. SU channel access probability for 10 SUs

length, but they had worse time to rendezvous than *RCCH-Sync*. In the same figures, we plot the theoretical results. One can see that the theoretical results are quite higher than the simulated value. The reason behind it is, when deriving this formula we assumed the

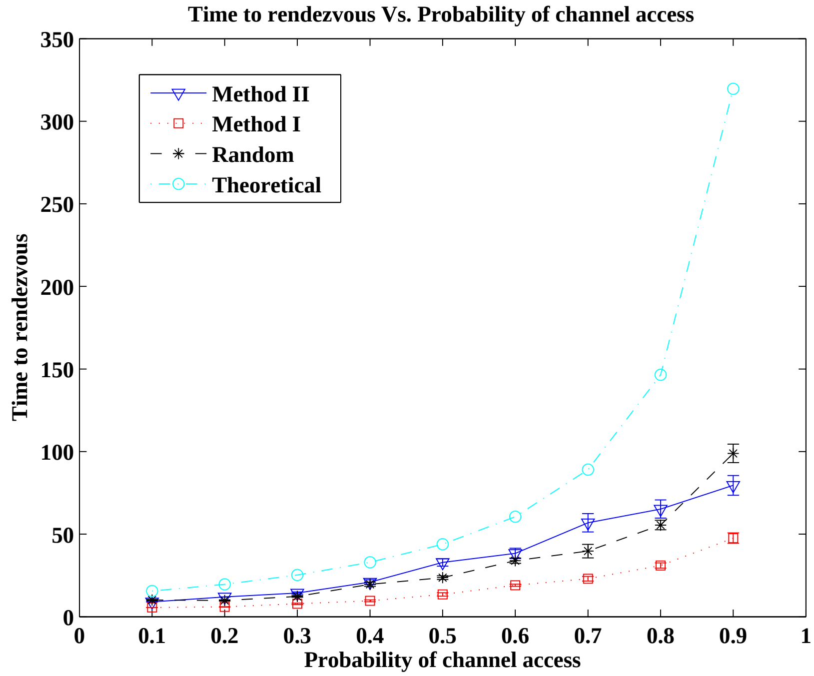

Figure 3.6.3: Average number of control slots Vs. SU channel access probability for 20 SUs

channels are accessed by the SUs according to the control channel selection scheme in all the time slots, while in the simulation, once the handshake is done the channels are accessed according to a different distribution which allows SUs to balance its load among all the channels due to the use of SU statistics in data phase channel selection. Furthermore, in the data phase the receiver is tuned to the channel the transmitter is using and it is not transmitting. Therefore, the receiver and the transmitter is synchronized to one another. Ideally, only the number of users in the handshake phase and the channels free from both PU activity and the SU data transmission activity should be used in this calculation. But due to the complex interactions of SUs this calculation was not feasible.

We used an exponential back-off scheme for the CSMA/CA access of the SUs. We limited the back-off exponent to 4 for the handshake phase and 3 for the Data phase. The time duration details for the control phase are given in Table 3.4. Since we use the same channels for both the handshake and the data phases the number of time slots for rendezvous is dependent on how aggressively the SUs access channels in the data phase. The back-off exponent of the control phase is set to be higher since the existing data transmissions should be given priority over the new transmissions.

Parameter	Value
Transmission rate	1Mbps
Sensing time per channel	0.05ms
Channel switch time	1μs
Slot duration	4.615ms
CW_{min}	15
Slot time	9μs
Propagation time	5μs

Table 3.4: Time duration for time slot

The time duration taken for the transmission of the RTS, CTS, ACK and DATA parts of the frame for a given data rate R, D bytes of payload and S bytes of channel sensing information are shown in equations (3.6.2), (3.6.3), (3.6.4) and (3.6.5). The expressions are taken from [86].

$$t_{RTS} = 20\mu s + \frac{(22 + (20 + S) \cdot 8)}{R}, \tag{3.6.2}$$

$$t_{CTS} = 20\mu s + \frac{(22 + (14 + S) \cdot 8)}{R}, \tag{3.6.3}$$

$$t_{ACK} = 20\mu s + \frac{(22 + 14 \cdot 8)}{R}, \tag{3.6.4}$$

$$t_{data} = 20\mu s + \frac{(22 + (28 + D) \cdot 8)}{R}. \tag{3.6.5}$$

3.6.2 Data Phase

In the simulations, once the initial handshake is done, each time slot was divided into 3 sub-slots. This number is equal to the number of channel switchings allowed at each time slot. The probability with which an SU begins a transmission was distributed according to the Bernoulli distribution with $\alpha = \{0.1, 0.2, \cdots 0.9\}$ being the probability of success. Then as long as the number of packets to be transmitted is above zero, packets were added to the transmission queue at the beginning of each main time slot according to the discrete uniform distribution having the set of outcomes $\{1, 2, 3\}$. The transmission was assumed to be complete when the queue gets empty. Then the probability of winning the contention was calculated for each SU by taking the ratio between the number time slots with successful packet transmission and the number of transmission attempts made, for the channels used. For the channels a given SU did not use, we assumed each channel i is used with probability $\frac{\alpha p_i}{\sum_{i=1}^{N} p_i}$ by other SUs as explained in Section 3.5.1.

We used the PST scheme to predict the channel state, for schemes Method-I and Method-II. In the Random scheme, the channels were arranged in each time slot according to the descending order of the probability of the channel begin free from PU activity. In the random scheme we used a scheme based on the hidden Markov models [57] to predict the channel status. In the simulation setup, an SU who acquired the channel, can transmit a packet in each sub-slot only if the PU is not present. If an SU transmitted when a PU is present, it is called an interfered sub-slot. This performance measure is plotted against the probability of SU channel access in figures 3.6.4, 3.6.5 and 3.6.6 for 5, 10 and 20 SUs respectively.

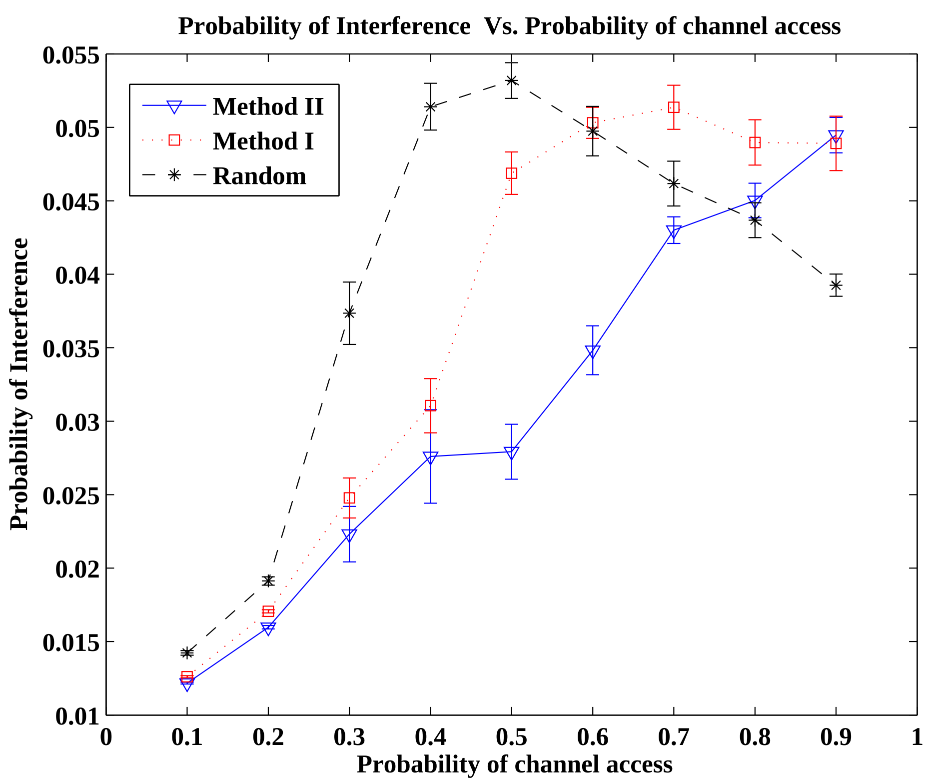

Figure 3.6.4: Probability of interference to PU Vs. SU channel access probability for 5 SUs

In Figure 3.6.4 we can see that Method-II has the least interference for most cases while Method-I lies between the Random graph and Method-II. The crossover between the Graph for Method-II and the Random scheme appears some where at the probability of channel access between 0.7 and 0.8. The trend of the graph is as expected. When the number of attempts increase, the probability of interference also increases.

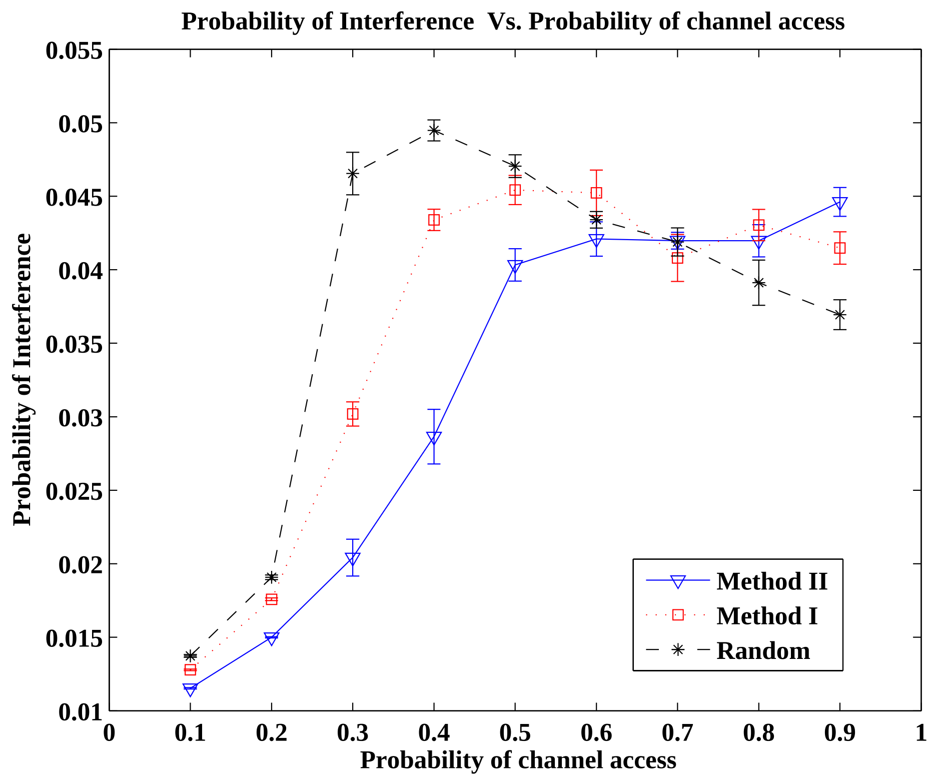

Figure 3.6.5: Probability of interference to PU Vs. SU channel access probability for 10 SUs

Similar to Figure 3.6.4, in Figure 3.6.5, we can see that Method-I has least interference for all SU channel access probabilities from 0.1 to 0.7. But the probability of interference to PU is less than 0.05 in both Method-I and Method-II. The Random method has better interference for channel access probabilities from 0.8 and 0.9. A possible reason behind this is, due to the usage of the SU statistics based on past contention failures, a given SU over-estimates the level of competition for the highly available channels and try to aggressively use the channels with high PU activity there by interfering the PU more. In the Random channel set selection, a channel with channel utilization 0.2 gets selected 50% of the time , since 5 out of 10 channel have this channel utilization. Fine tuning of the Data collection window of contention failures and successes should be able to keep the level of interference to the PUs lower in both Method-I and Method-II.

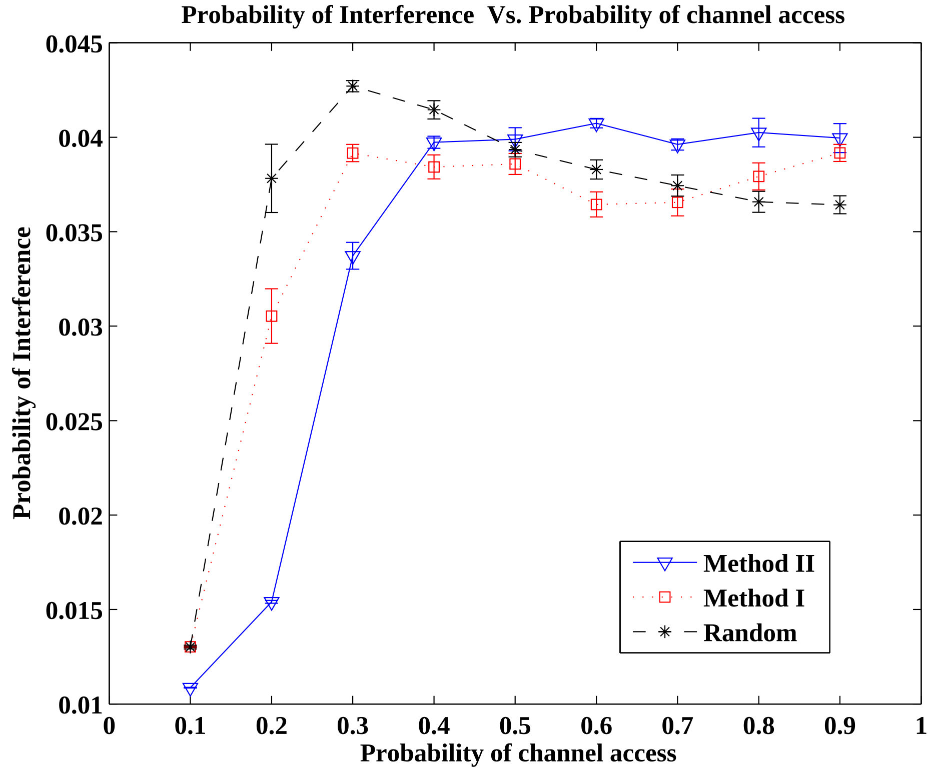

Figure 3.6.6: Probability of interference to PU Vs. SU channel access probability for 20 SUs

In Figure 3.6.6 also we can see a similar trend as in Figure 3.6.5.

The graphs depicting the fluctuation of the probability of successful transmission is shown in figures 3.6.7, 3.6.8 and 3.6.9, which are plotted for 5, 10 and 20 SUs respectively.

Figure 3.6.7 shows the probability of successful transmission for 5 SUs. Here, we can see that Method-II performs best, while Method-I performs better than the Random scheme for most of the cases.

Similarly, in Figure 3.6.8, we can see that method-II performs better than the Random scheme for all SU channel access probabilities except at 0.9. When the number of SUs was increased to 20, Method-II produced better throughput for all the cases except for probability of channel access from 0.5 to 0.7, as shown in Figure 3.6.9. Therefore, we can say that Method-II has the best performance. The probability of successful transmission depends on a lot of factors. One of the prominent factors are the probability of successfully

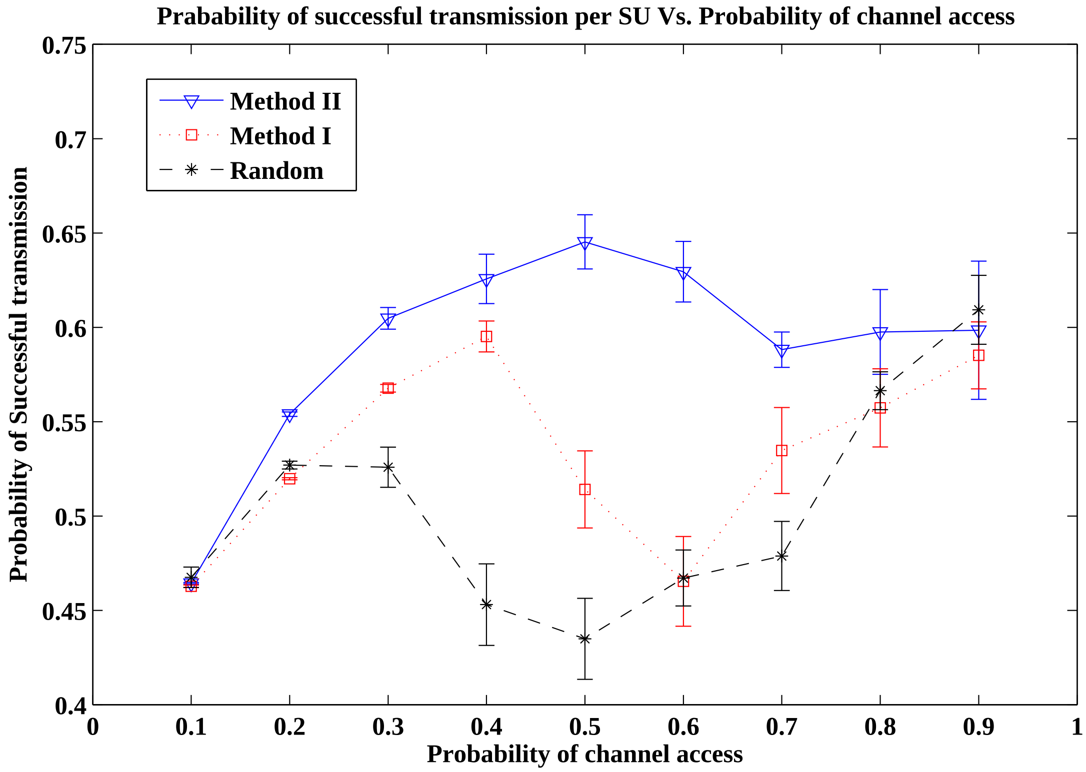

Figure 3.6.7: Probability of succesful transmission per SU Vs. SU channel access probability for 5 SUs

predicting the state of the channel. The accuracy of prediction depends on the channel state history the SUs posses. In the Random scheme since the SU picks channels at random, the channel state information data of past slots from one transmission to the next becomes useless since the probability of choosing a completely different set of channels is high. In Method-I and II this probability is lower than the Random scheme. This can be a cause of Random channel selection scheme performing poorly.

In figures 3.6.10, 3.6.11 and 3.6.12, we plot the probability of collision among SUs for the cases 5, 10 and 20 SUs respectively. A collision happens when two or more SUs transmit within the vulnerable time. Then in figures 3.6.13, 3.6.14 and 3.6.15, we plot the average number of SUs concurrently using the same channel, which is a measure of channel load. We calculated this performance measure comparing the arrangement of channels of the SUs. For example, if we have two SUs in the network and user 1 has a channel arrangement $\langle 1, 5, 6\rangle$ and user 2 has $\langle 2, 5, 6\rangle$ the average number of common channels used is $\frac{1+1+2+2}{10} = 0.6$. If one compares the respective graphs, it is possible to see that, in

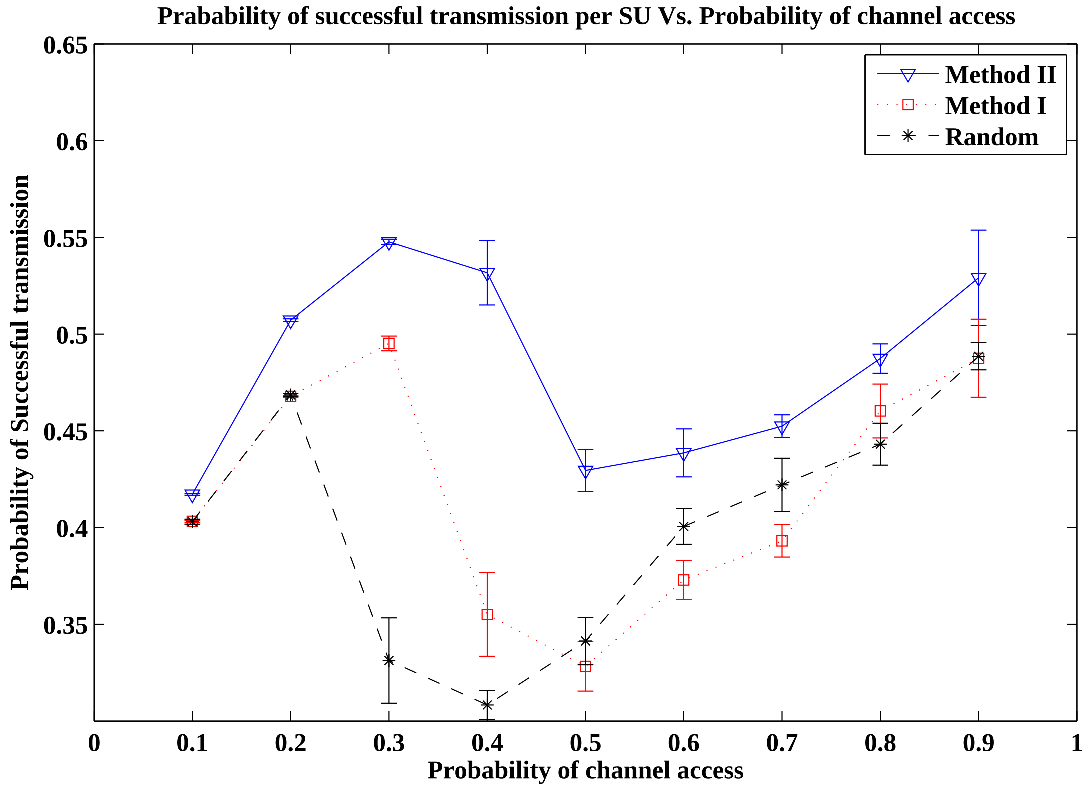

Figure 3.6.8: Probability of succesful transmission per SU Vs. SU channel access probability for 10 SUs

Method-II, less SUs use the same channel for most channel access probabilities. Therefore, this leads to less collision probability.

3.7 Conclusion

In this chapter we presented a proactive channel access MAC for an Ad-Hoc CR network having a REM for initial set up, and showed how this MAC could be used to keep the interference experienced by the PUs at a low level while providing better throughput for the SUs. The novel feature of this research was the incorporation of the SU statistics when selecting a channel set to be used. Furthermore, we introduced a SH CCC and derived the theoretical properties of it. This CCC used PU channel access probabilities when selecting the hopping pattern in the control phase. Then, in the data transmission phase, we used a prediction scheme called the PST algorithm, which has a very low training and memory complexity. We tested our schemes Method-I and Method-II in a scenario with a MAP based chan-

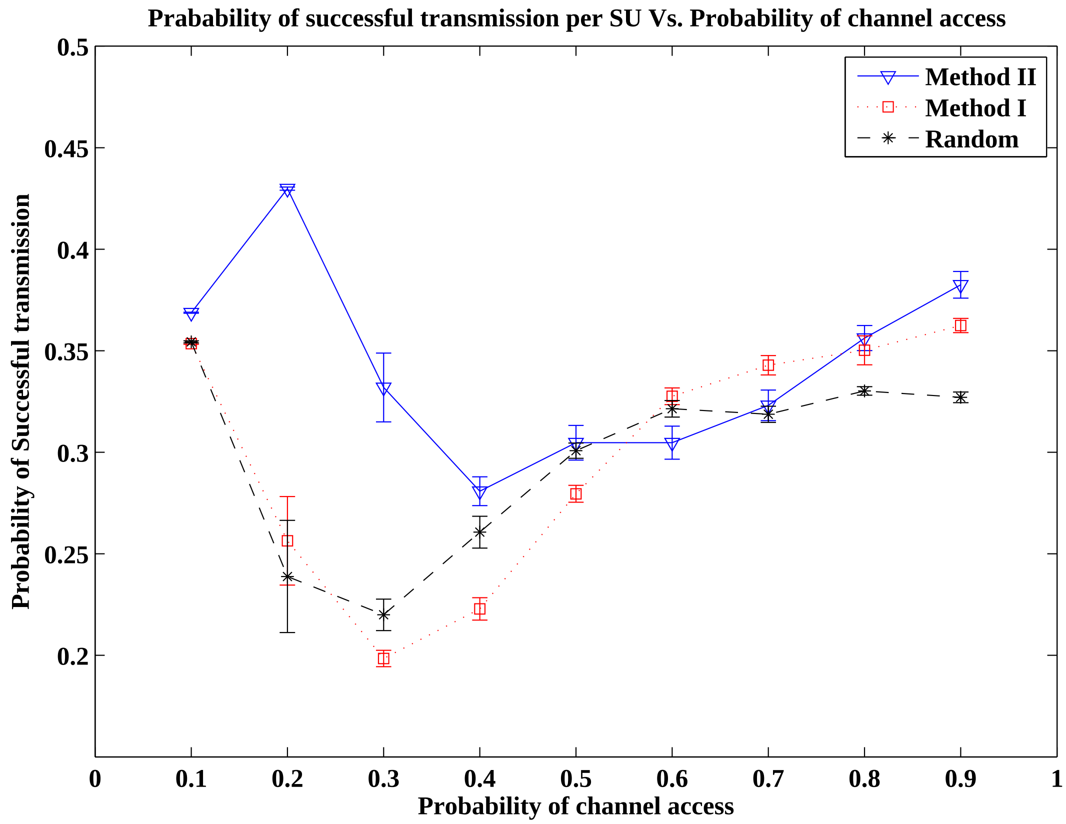

Figure 3.6.9: Probability of succesful transmission per SU Vs. SU channel access probability for 20 SUs

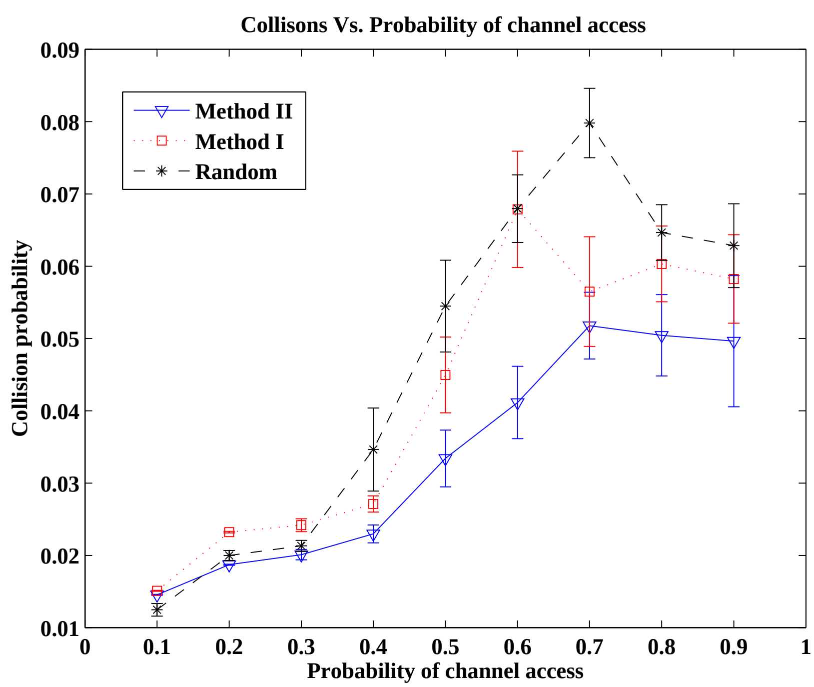

Figure 3.6.10: Probability of collision amoung SUs Vs. SU channel access probability for 5 SUs

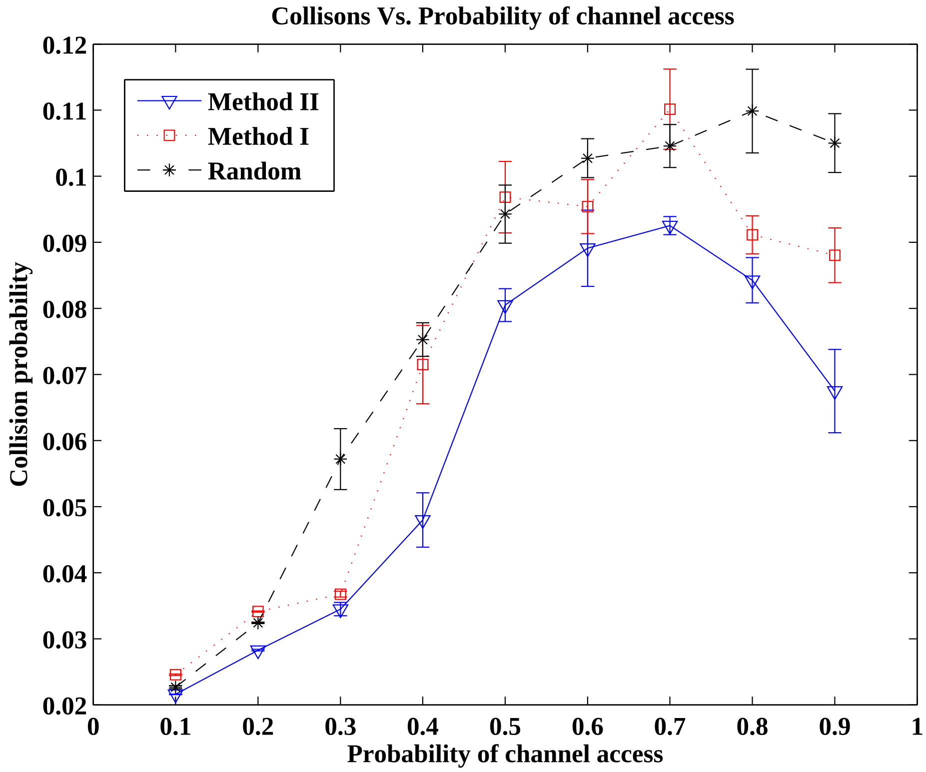

Figure 3.6.11: Probability of collision amoung SUs Vs. SU channel access probability for 10 SUs

nel state distribution with different channel utilization levels with imperfect sensing, and showed the impact the channel set selection method has on the throughput and the interference to the PUs. In the simulations, our channel set selection scheme Method-II had better probability of successful transmission than the random channel set selection scheme, while having a PU interference probability less than 0.05. When comparing the Method-I and Method-II in terms of complexity, Method-I has lower complexity than Method-II. But, Method-I had a smaller probability of successful transmission than Method-II. The figures showing the probability of interference and the probability of successful transmission combined give an idea of the accuracy of prediction. Higher the prediction accuracy lower will be the interference. But one should consider the probability of transmission as well when evaluating the accuracy of prediction. Because one can achieve a null probability of interference if one does not transmit. Therefore, the scheme with the maximum probability of transmission for a given maximum probability of interference has the best prediction capability.So when considering all of these factors we can conclude that our MAC scheme is

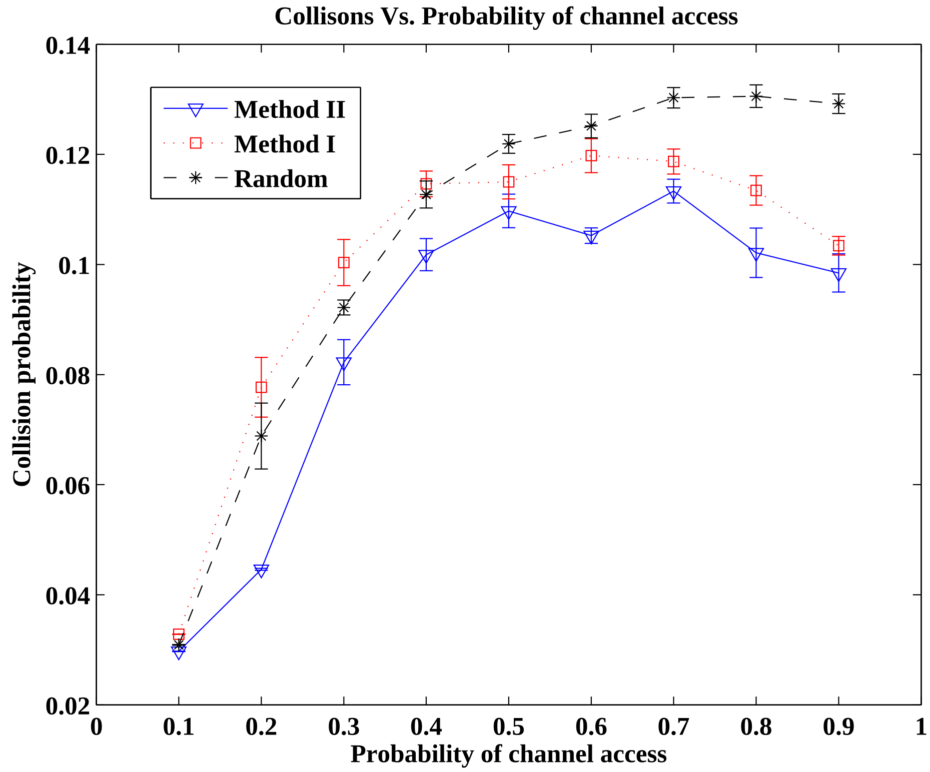

Figure 3.6.12: Probability of collision amoung SUs Vs. SU channel access probability for 20 SUs

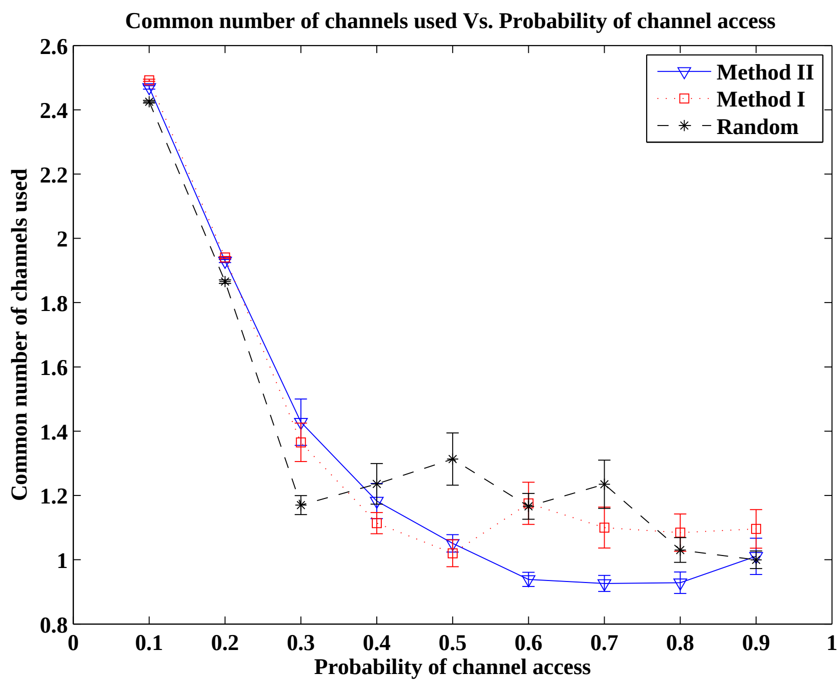

Figure 3.6.13: Average number of channels simultaneously used by SUs Vs. SU channel access probability for 5 SUs

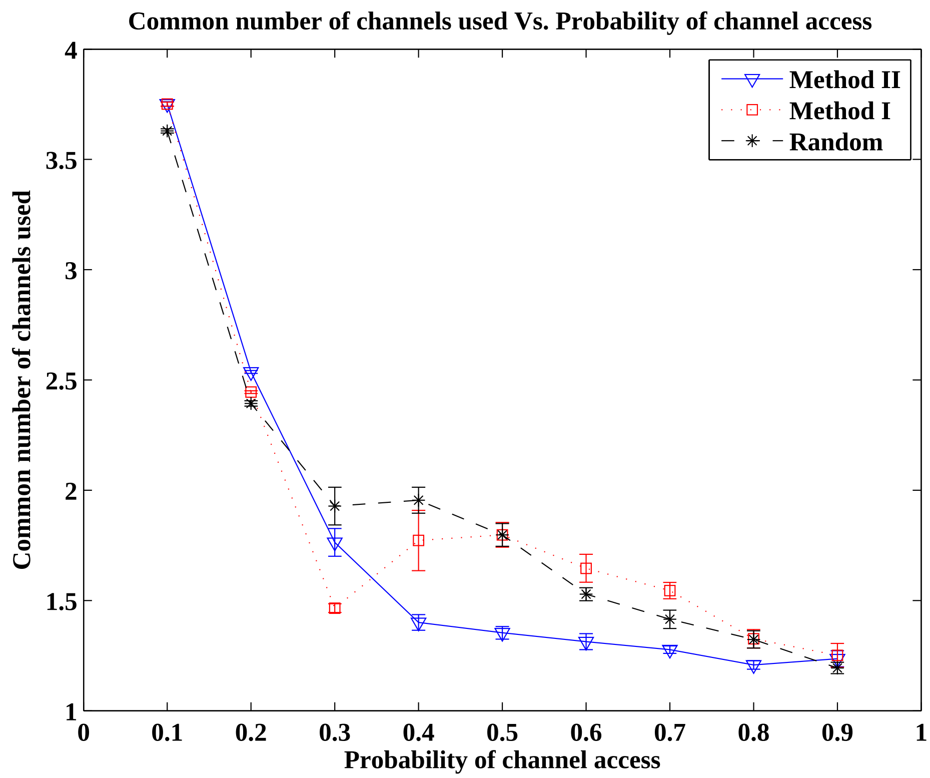

Figure 3.6.14: Average number of channels simultaneously used by SUs Vs. SU channel access probability for 10 SUs

candidate worthy of consideration in proactive channel access in an Ad-Hoc setting similar to the one the schemes were tested on.

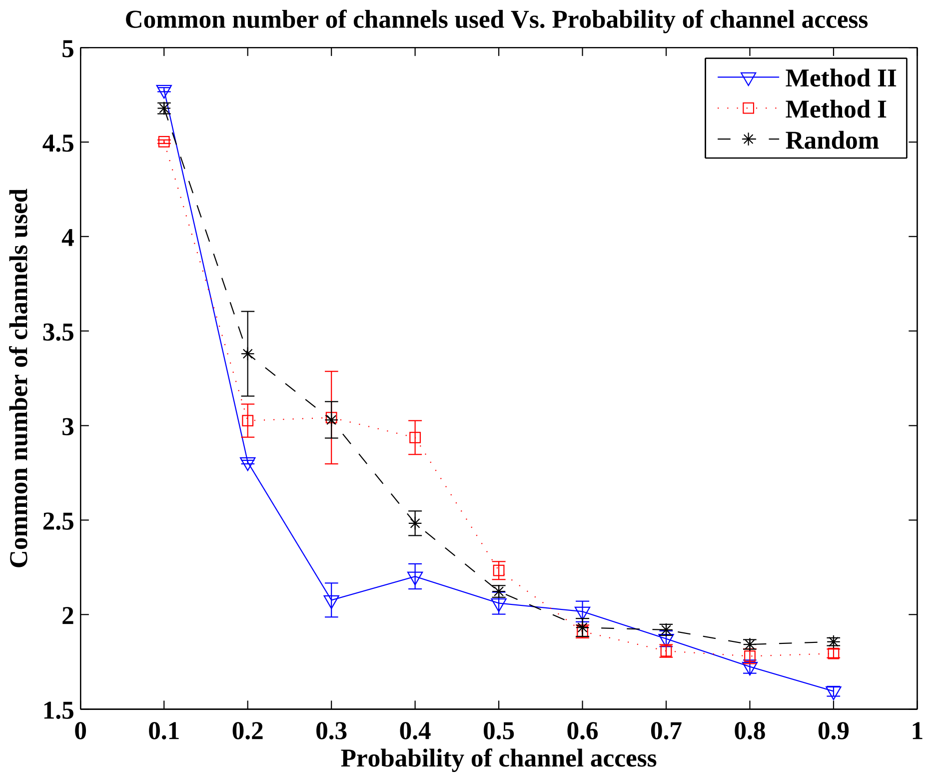

Figure 3.6.15: Average number of channels simultaneously used by SUs Vs. SU channel access probability for 20 SUs

Chapter 4

Network Assisted Underlay Protocol for Bi-directional FD CRs with Interference Estimation

In this chapter, we propose a network assisted joint channel assignment and power allocation scheme for a device-to -device (D2D) CR network, which makes use of FD communication. In this scheme, each pair of SUs decide which channels should be used as half duplex (HD) and which channels should be used as FD to maximize the throughput between the pair. While maximizing the throughput, the interference to the primary users (PUs) is kept below a threshold by means of controlling the interference at monitoring stations. Monitoring stations (MSs) inform the primary base station the level of interference which was present at the previous measuring instant, using either a control channel or over the Internet as in the case of small cell networks. The term MSs was used to keep the scheme as general as possible. Depending on the network setup this MS can be a CR, a dedicated interference sensor, a PU base station or a PU. We assume that the CRs register themselves with the primary base station and thus are synchronized with the primary network. Furthermore, the network assistance improves the energy efficiency providing assistance on device discovery and security [90]. Device discovery is the process which enables one CR device to find the intended receiver and to find out whether D2D communication is possible. We call the network setup used in this chapter a cognitive D2D setup, since this setup is different from the LTE-advanced standard where the base station schedules D2D access using dedicated resources. This concept of a MS was first used in [49]. In this scheme the CR pairs make their transmission decisions using the interference information at the MS and the information on the channel gains between the pair and the monitoring stations, after solving an optimization problem. This protocol needs information of the number of active CR pairs in the network and the interference level on all the channels at each MS at the start of each time slot. Therefore, the control overhead of broadcasting interference information every time slot, registering users and finding if they are active is high. To reduce this overhead we proposed an estimation algorithm for the interference at the MSs based on the local interference observed. In this second algorithm we use least squares estimation and Kalman filtering. This second algorithm does not need the information on the number of active users. Furthermore, it can estimate the interference into the future, but the quality reduces with the projection interval because the parameter estimation error becomes high. Therefore, if the MSs broadcast information on interference every τ time slots, it should broadcast the information on past τ time slots. This scheme is discussed in Section 4.5.

4.1 System Model

In the system we considered, there are multiple pairs of secondary users/CR pairs (SUs/CRs) who are capable of performing device-to-device communication between themselves in a full duplex manner. These CR pairs are assumed to be within the coverage area of a base-station. Furthermore, these CR pairs register with the base station and are synchronized with it. In addition to that, there are several monitoring stations which monitor the interference caused by the SUs. We assume that the MS is capable of distinguishing between the SU signals and the PU signals. As mentioned in [49], this can be achieved by enforcing a universally quite period for the SU pairs. As mentioned before, we assume that the MSs

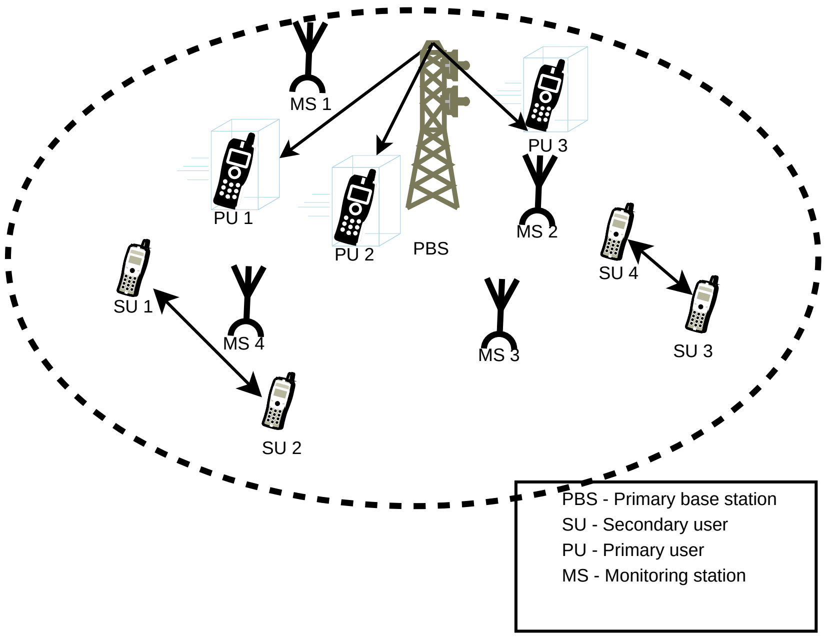

Figure 4.1.1: Network topology of the proposed scheme

report to the primary base station the level of interference which were present at the previous measuring instant, using either a control channel or over the Internet as in the case of small cell networks. Then, the base station broadcast this information together with the number of pairs of active users to the CR pairs, using a control channel at the start of the next time slot. The decisions on the power level and the channels to be used are taken by each pair considering the interference from other SUs and the interference caused at the MSs by all the SUs. Since one of the members in each pair takes its own decision about the channel access, only a limited amount of control information is exchanged between the two members. We designed our optimization problem such that the channel assignment can choose some channels to be half duplex (HD) while some others are full duplex (FD), which provided extra flexibility. A diagram of the system with two pairs of users is shown in Figure 4.1.1.

The channel assignment and power allocation decision of each pair of users only depends on the interference present on each channel at each user of the pair, the last informed SU interference observed at the MSs and the number of active SU pairs. Furthermore, since there is no coordination between the CR pairs, the gap between the informed interference and the threshold should be shared among the CR pairs. To this end, we propose each CR pair to draw a uniformly distributed random number δ in the range $[0, \min(1, \frac{2}{M})]$, where M is the number of active CR pairs in the network. We assume that the value of M is also transmitted by the base-station when transmitting the data on the current interference on channels. This δ is the fraction of the available gap usable by the CR pair at that time slot. Then, to reduce the overhead of maintaining the list of active users we propose an interference estimation scheme which estimates the interference at the MSs based on the local interference.

4.1.1 Problem Formulation

Using the above mentioned assumptions, we can formulate an optimization problem to find the best channel assignment and power allocation for all pairs of SUs to maximize the achievable sum data rate of all active pairs, as shown in Equation (4.1.1).

$$
\begin{aligned}
&\max_{\mathbf{x},\mathbf{p}} \sum_{k=1}^{M}\sum_{i=1}^{N}\sum_{\theta=\{1,2\}} x_{ik}^{-\theta}\log_2\left(1+\frac{p_{ik}^{-\theta}g_s^k}{I_{ik}^{\theta}+\sum_{j=1,j\neq k}^{M}\sum_{\zeta=1}^{2}p_{ij}^{\zeta}h_{jk}^{\zeta\theta}+p_{ik}^{\theta}h_k^{\theta}\alpha}\right), && \text{(P 4.1.1)}\\
&\text{s.t.} \sum_{i=1}^{N} x_{ik}^{\theta} \geq 1 \quad \forall\theta=\{1,2\},\ k=\{1,2,...,M\}, && \text{(C1)}\\
&\sum_{i=1}^{n} p_{ik}^{\theta} \leq P_{max} \quad \forall\theta=\{1,2\}\ k=\{1,2,...,M\}, && \text{(C2)}\\
&\frac{p_{ik}^{-\theta}g_s^k}{I_{ik}^{\theta}+\sum_{j=1,j\neq k}^{M}\sum_{\zeta=1}^{2}p_{ij}^{\zeta}h_{jk}^{\zeta\theta}+p_{ik}^{\theta}h_k^{\theta}\alpha} \geq x_{ik}^{-\theta}\gamma_{min} \quad \forall\theta=\{1,2\}, i=\{1,2,...,N\}, &&\\
&k=\{1,2,...,M\}, && \text{(C3)}\\
&\sum_{j=1}^{M}\sum_{\theta=\{1,2\}} x_{ij}^{\theta}p_{ij}^{\theta}g_{jm}^{\theta} \leq \Gamma_i \quad \forall i=\{1,2,...,N\},\ \forall m=\{1,...,R\}. && \text{(C4)}
\end{aligned}
\tag{4.1.1}
$$

We define the symbols used in Equation (4.1.1) in Table 4.1.

In optimization problem formulation (4.1.1), we maximize the sum data rate of the CR user pairs. Here, we assume that the device discovery is already done and the pairs of users

Symbol	Definition
x_{ik}^{θ}	Usage indicator for channel i at user θ of pair k
p_{ik}^{θ}	Power used on channel i at user θ of pair k
g_s^k	Channel gain between the users of pair k
I_{ik}^{θ}	Sum of cellular user interference and Gaussian noise for channel i at user θ of pair k
h_k^{θ}	Channel gain between the transmitter radio chain and the receiver radio chain of user θ of pair k
$h_{jk}^{\zeta\theta}$	Channel gain between the transmitter ζ of pair j and the receiver θ of pair k
α	Self-interference attenuation parameter
g_{km}^{θ}	Channel gain between the MS m and user θ of pair k

Table 4.1: Symbol definitions

are already formed. The CR user pairs are indexed with k and j in (4.1.1). We assume the number of active D2D pairs to be M. We use the indices θ and ζ to identify the users in a given D2D pair. We use the negative index to make the optimization problem formulation compact. For example, if two users having indices 1 and 2 are communicating with each other, then if $\theta = 1$ then $-\theta = 2$ and vice versa. The channels in this formulation are indexed with i. We assume there are N channels in total. The objective function (P 4.1.1) is formulated to maximize the sum data rate of the M D2D pairs, where x_{ik}^{θ} and p_{ik}^{θ} are the channel assignment indicator and power allocation respectively. x_{ik}^{θ} is an indicator function, which takes the value 1 when channel i is used by member θ of pair k, and zero otherwise. p_{ik}^{θ} is the amount of power allocated by member θ of pair k on channel i. The mode selection between FD and HD are achieved using the indicator function x_{ik}^{θ} according to the formula shown in Equation (4.1.2). Here, $mode_{ik}$ is the transmission mode of user pair k on channel i, $CR1$ is the user 1 of a given pair and $CR2$ is the user 2 of a given pair.

$$mode_{ik} = \begin{cases} FD & if\ x_{ik}^1 = x_{ik}^2 = 1 \\ HD,\ CR1 \rightarrow CR2 & if\ x_{ik}^1 = 1\ and\ x_{ik}^2 = 0 \\ HD,\ CR2 \rightarrow CR1 & if\ x_{ik}^2 = 1\ and\ x_{ik}^1 = 0 \\ No\,Transmission & if\ x_{ik}^2 = 0\ and\ x_{ik}^1 = 0 \end{cases} \tag{4.1.2}$$

The expression $\sum_{j=1,j\neq k}^{M} \sum_{\zeta=1}^{2} p_{ij}^{\zeta} h_{jk}^{\zeta\theta}$, in both objective function (P 4.1.1) and the constraint (C3) is the interference caused by other D2D pairs on user θ of pair k. Then, the self-interference at user θ of pair k is calculated using the expression $p_{ik}^{\theta} h_k^{\theta} \alpha$.

Constraint (C1) makes sure each member of all the pairs gets at least a single channel to transmit. Constraint (C2) limits the total power that can be utilized by each member to P_{max}. Constraint (C3) makes sure the signal to noise plus interference ratio (SINR) on channel i at the receiver θ of the pair k is greater than γ_{min} when the channel is used by member $-\theta$ (transmitter) of pair k. The final constraint (C4) makes sure the total interference from all the D2D pairs on channel i at the monitoring station is less than the threshold Γ_i.

The problem described above is a mixed integer non-linear program, which is not solvable in polynomial time. Even if we made simplifying adjustments to solve this problem sub-optimally, the amount of data that should be sent to the centralized controller increases the control overhead. Because of these reasons, we proposed to solve this problem in a decentralized manner, with minimum network assistance. When examining the optimization problem (4.1.1), we can see that objective function (P 4.1.1), and the two constraints (C3) and (C4) has some terms which introduces coupling between the CR user pairs. If we try to solve this problem using primal decomposition as in [91], it requires a significant amount of message passing among the CR user pairs. Therefore, we use the previous time slot measurements of these coupling terms in order to keep the control overhead to a minimum.

One of the coupling terms in this formulation is the term $\sum_{j=1, j \neq k}^{M} \sum_{\zeta=1}^{2} p_{ij}^{\zeta} h_{jk}^{\zeta\theta}$, which is the total CR interference felt by user θ of pair k. The sum of total interference constituting of the white noise, PU interference and the CR interference can be measured at each time slot. But if the all the CR users transmit in a synchronized manner in each time slot this measurement give only the sum of interference of PUs and the noise. Therefore, we use the interference measurement in the previous time slot. Let this total interference on channel i from PUs and other SUs at user θ of pair k be represented by $\tilde{I}_{ik}^{\theta}$. The second coupling term is $\sum_{j=1, j \neq k}^{M} \sum_{\theta=\{1,2\}} x_{ij}^{\theta} p_{ij}^{\theta} g_{jm}^{\theta}$. We use the past slot interference measure to remove this coupling term. Measuring this interference term at a place other than the

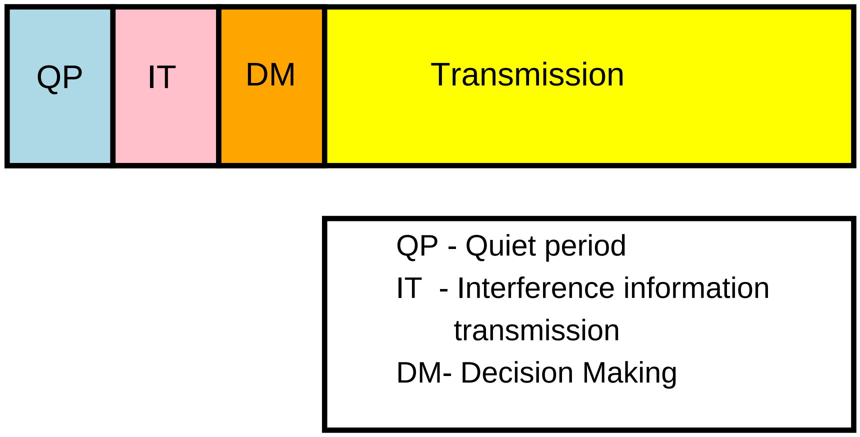

Figure 4.1.2: Slot structure of D2D CRs

PU receiver requires us to make use of a quiet period as shown in the time slot structure in Figure 4.1.2. In this time slot structure for the D2D CR pairs, the first sub-slot is a quiet period for the CRs. Then, the monitoring stations make measurements of the PU interference on the channels and then subtract that from the total interference to measure the CR interference. Afterwards, these interference measures are sent to the primary base station and this information subsequently gets transmitted to the CRs in the sub-slot IT. This interference information contains the interference measures in the previous time slot. Then, to find the interference caused by the other CRs at the monitoring station m on channel i, we subtract the interference caused by the CR pair k in the previous time slot at that channel at the monitoring station. Let this term be represented by J_{im}.

Having these interference parameters $\tilde{I}^{\theta}_{ik}$ and J_{im}, we decompose the problem in (4.1.1) into M problems. Each one of these problems are solved by a given pair of CRs in parallel. We introduced two additional notations in the decomposed problem (4.1.3) and dropped the subscript k from both x^{θ}_{ik} and p^{θ}_{ik} . The first addition is the introduction of $[\cdot]^{+} = \max(0, \cdot)$ in constraint (C4) of (4.1.3). The second is the introduction of δ in constraint (C4) of (4.1.3). There is a possibility for J_{im} to be higher than Γ_i because all the CR pairs make decisions in a decentralized manner. For this reason, we introduced the $[\cdot]^{+}$ operator so that the problem stays feasible. Then there is a possibility that the users who joined the network at less busy times holding on to the same amount of resources in the busy times.

Because of this, the new comers to the network in busy times can have a smaller data rate unfairly. To avoid this situation we introduced the parameter δ which is a randomly generated number for a given time slot. This random number δ has a uniform distribution in the range $[0, \min(1, \frac{2}{M})]$, where M is the number of active CR pairs in the network. This random fraction makes sure one CR pair does not hold onto a constant amount of resources, and only a random fraction of the available interference gap between the threshold and the actual interference at monitoring stations is used by a given CR pair.

$$
\begin{aligned}
&\max_{\mathbf{x},\mathbf{p}} \sum_{i=1}^{N} \sum_{\theta=\{1,2\}} x_i^{-\theta} \log_2\left(1 + \frac{p_i^{-\theta} g_s}{\tilde{I}_i^{\theta} + p_i^{\theta} h_s \alpha}\right) , && \text{(P 4.1.3)} \\
&\text{s.t.} \sum_{i=1}^{N} x_i^{\theta} \geq 1 \quad \forall\theta = \{1,2\} , && \text{(C1)} \\
&\sum_{i=1}^{n} p_i^{\theta} \leq P_{max} \quad \forall\theta = \{1,2\} , && \text{(C2)} \\
&\frac{p_i^{-\theta} g_s}{\tilde{I}_i^{\theta} + p_i^{\theta} h_s \alpha} \geq x_i^{-\theta} \gamma_{min} \quad \forall\theta = \{1,2\}, i = \{1,2,...,N\} , && \text{(C3)} \\
&\sum_{\theta=\{1,2\}} x_i^{\theta} p_i^{\theta} g_m^{\theta} \leq \delta[\Gamma_i - J_{im}]^+ \quad \forall i = \{1,2,...,N\}, && \text{(C4)} \\
&\quad \forall m = \{1,...,R\} .
\end{aligned}
\tag{4.1.3}
$$

4.1.2 Problem Decomposition

The above problem is also a mixed integer non-linear program. But, this problem has less number of decision variables. Solving this problem within polynomial time in its current form is impossible. Therefore, we decomposed the problem into two sub-problems, one dealing with power allocation, and the other dealing with channel assignment. In the decomposition, we assumed an initial feasible solution, $P_i^{\theta} \quad \forall i = \{1,2,...,N\}, \theta = \{1,2\}$, exists for the power allocation and first solved the channel assignment problem (4.1.4). We have substituted p_i^{θ} in Problem (4.1.4) with the initial feasible solution P_i^{θ} when deriving Problem (4.1.3). Furthermore, the Constraint (C2) of Problem (4.1.3) is not present here, since it has nothing to do with the channel assignment. The method of solving this problem is explained in Section 4.2.

$$\begin{aligned}
&\max_{\mathbf{x}} \sum_{i=1}^{N} \sum_{\theta=\{1,2\}} x_i^{-\theta} \log_2 \left(1 + \frac{P_i^{-\theta} g_s}{\tilde{I}_i^{\theta} + P_i^{\theta} h_s \alpha} \right) , \\
&\text{s.t.} \sum_{i=1}^{N} x_i^{\theta} \geq 1 \quad \forall \theta = \{1,2\} , \\
&\frac{P_i^{-\theta} g_s}{\tilde{I}_i^{\theta} + P_i^{\theta} h_s \alpha} \geq x_i^{-\theta} \gamma_{min} \quad \forall \theta = \{1,2\}, i = \{1,2,...,N\} , \\
&\sum_{\theta=\{1,2\}} x_i^{\theta} P_i^{\theta} g_m^{\theta} \leq \delta [\Gamma_i - J_{im}]^+ \quad \forall i = \{1,2,...,N\}, \\
&\quad \forall m = \{1,...,R\} .
\end{aligned} \tag{4.1.4}$$

After we find a solution, $X_i^{\theta} \quad \forall i = \{1,2,...,N\}, \theta = \{1,2\}$, for the problem (4.1.4) , we substitute X_i^{θ}, $\forall i = \{1,2,...,N\}, \theta = \{1,2\}$, in the optimization problem (4.1.5) to find the optimum power allocation. We have used the logarithmic property $\sum_{i=1}^{N} \log(A_i) = \log(\prod_{i=1}^{N} A_i)$ and the property $n \log(A) = \log(A^n)$, when deriving the objective function (P 4.1.5) from (P 4.1.3). Here, we have substituted x_i^{θ} in Problem (4.1.3) with the solution X_i^{θ} we got after solving Problem (4.1.4). Furthermore, we have removed Constraint (C1) in Problem (4.1.3).

$$\begin{aligned}
&\max_{\mathbf{p}} \log_2 \left(\prod_{i=1}^{N} \prod_{\theta=\{1,2\}} \left(1 + \frac{p_i^{-\theta} g_s}{\tilde{I}_i^{\theta} + p_i^{\theta} h_s \alpha} \right)^{X_i^{-\theta}} \right) , \qquad \text{(P 4.1.5)} \\
&\text{s.t.} \sum_{i=1}^{n} p_i^{\theta} \leq P_{max} \quad \forall \theta = \{1,2\} , \\
&\frac{p_i^{-\theta} g_s}{\tilde{I}_i^{\theta} + p_i^{\theta} h_s \alpha} \geq X_i^{-\theta} \gamma_{min} \quad \forall \theta = \{1,2\}, i = \{1,2,...,N\} , \\
&\sum_{\theta=\{1,2\}} X_i^{\theta} p_i^{\theta} g_m^{\theta} \leq \delta [\Gamma_i - J_{im}]^+ \quad \forall i = \{1,2,...,N\}, \\
&\quad \forall m = \{1,...,R\} .
\end{aligned} \tag{4.1.5}$$

We discuss the solution methodology in the next Section.

4.2 Solution Methodology

In the outlined procedure above, we needed an initial solution for the power allocation to get an initial solution to start with. First, we assumed that all the channels are used. Then, we calculated the required power level to satisfy the minimum SINR constraint. To calculate this power, we solved the system of equations shown in Equation (4.2.1). The system has a unique solution since the interference term $\tilde{I}_i^{\theta}$ is independent from one (i, θ) pair to another.

$$\frac{p_i^{-\theta} g_s}{\tilde{I}_i^{\theta} + p_i^{\theta} h_s \alpha} = \gamma_{min} \quad \forall \theta = \{1, 2\} \quad i = \{1, 2, ..., N\} \tag{4.2.1}$$

As one can see, the Optimization problem (4.1.4) is a linear binary optimization problem which can be solved using the branch and bound algorithm. We used the solution to Problem (4.2.1) as the initial solution $P_i^{\theta} \quad \forall i = \{1, 2, ..., N\}, \theta = \{1, 2\}$. Then, we solved Problem (4.1.4) using the branch and bound algorithm. Let the solution to this problem be X_i^{θ}, $\forall i = \{1, 2, ..., N\}, \theta = \{1, 2\}$. Having this solution in hand,we focus on the power allocation problem. In the power allocation problem, we have a non-convex continuous optimization problem. Although it is not convex, we can convert it into a fractional programming problem, which is a Monotonic optimization problem [92]. For a given channel allocation X_i^{θ}, the optimization problem can be formulated as shown in Equation (4.1.5).

Due to the monotonicity of the $\log_2(\cdot)$ function in Equation (4.1.5), we can remove the logarithm and find the solution for the problem using fractional programming [92] or signomial programming [93]. According to [94], the monotonic optimization is capable of providing a better solution than the signomial programming approach. We used a modified version of the monotonic optimization frame work used in [94], which is known to have better convergence as shown in [92] and [95].

As one can see, all the MSs are involved in the last constraint of the optimization problem given in Equation (4.1.5). But, if we choose the MS having the minimum ratio of interference gap to the maximum channel gain, between the user and MS, we only have to consider one MS for each channel. The index of this MS was found mathematically using Equation (4.2.2) for channel i, where $i \in \{1, 2, ..., N\}$.

$$m_i^* = arg \min_{m\in\{1,...,R\}} \left\{ \frac{\delta(\Gamma_i - J_{im})}{max_\theta\{g_m^\theta\}} \right\} . \tag{4.2.2}$$

The optimization method carried out for the optimal power allocation for a given X_i^θ is described in Section 4.3. After we obtain a solution, P_i^θ, for the power allocation, we substitute that solution into the channel assignment problem again. This procedure is carried out until the absolute difference between the solutions to Problem (4.1.5) in two consecutive iterations, is below a threshold ϵ. A similar alternating approach between the channel assignment and power allocation to solve joint allocations was proposed in [11]. Algorithm 3 outlines this alternating optimization procedure.

Algorithm 3 Joint Optimization

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1. **Input:** $g_s,\ \tilde{I}_i^\theta,\ h_s,\ \alpha,\ \Gamma_i,\ J_{im},\ P_{max}, g_m^\theta, g_s\gamma_{min},\ \epsilon,\ \forall\theta = \{1, 2\}\ i = \{1, 2, ..., N\}$
2. **Output:** $X_i^\theta,\ P_i^\theta$
3. Initialize: $\{P_i^\theta\} \Leftarrow Solve$ (4.2.1)
4. Initialize: $\{\tilde{P}_i^\theta\} \Leftarrow Solve$ (4.3.1)
5. **while** $|\{\tilde{P}_i^\theta\} - \{P_i^\theta\}| \geq \epsilon$ **do**
6. $\quad \{X_i^\theta\} \Leftarrow Solve\ (4.1.4)\ with\ input\{P_i^\theta\}$
7. $\quad \{\tilde{P}_i^\theta\} = \{P_i^\theta\}$
8. $\quad \{P_i^\theta\} \Leftarrow Solve\ (4.1.5)\ with\ input\{X_i^\theta\}$
9. **end while**

4.3 Optimization Framework

Assume the solution of the set of equations (4.2.1) for the given X_i^θ, is P_i^θ. We used this P_i^θ to calculate the channel assignment solving the optimization problem shown in Equation (4.1.4). The solution to this problem was found using the standard technique, where the relaxed problem was solved first, and then the integer solutions were derived using the branch and bound technique. This pair of solutions X_i^θ and P_i^θ constituted the initial feasible solution, if a solution exists. If an initial solution does not exist this problem is infeasible.

After this initial solution is calculated, any vector $(q_1^1, q_2^1, \cdots, q_N^1, \cdots, q_N^2)$ strictly less than $(P_1^1, P_2^1, \cdots, P_N^1, \cdots, P_N^2)$ element-wise is not feasible for the given X_i^θ.

4.3.1 Power Allocation Algorithm

The power allocation algorithm used in this work is based on the monotonic optimization framework presented in [92]. In this framework, the feasible region was covered with hypercubes by generating lower and upper corners of the cubes. Then, the optimal solution happens to lie on the line connecting the lower and upper corners of one of the hyper cubes. Therefore, to start the algorithm we needed both an initial lower corner and an upper corner. We used the power allocation found using the system of equations (4.2.1), $\mathbf{P} = (P_1^1, P_2^1, \cdots, P_N^1, \cdots, P_N^2)$, to be the lower corner. In order to find the upper corner, we solved the simple set of equations shown in Equation 4.3.1 for all the (i, θ) pairs, where $X_i^\theta = 1$ otherwise $p_i^\theta = 0$.

$$X_i^\theta p_i^\theta = \delta min_m \left(\frac{[\Gamma_i - J_{im}]^+}{g_m^\theta} \right) \quad \forall i = \{1, 2, ..., N\} \tag{4.3.1}$$

After the generation of the initial hyper cube we can follow *Algorithm PA* in [92] to find the optimal power allocation. We are providing a brief overview of the algorithm for the convenience of the reader in Appendix B.

4.4 Complexity Analysis of the Optimization Scheme

The complexity of the proposed optimization framework can be analyzed in two parts. The first part is the complexity of the binary linear optimization scheme and the second part is the complexity of the monotonic optimization scheme. In both of these parts the complexity does not depend on the number of pairs of SUs in the system. This is a result of decomposing the original problem enabling each pair of users to individually select channels and the power level. However, the complexity of the scheme depends on the number of channels used. Since this scheme is an iterative scheme we analyze the complexity of performing the operations in one iteration. According to [92], this polyblock based algorithm stops in finitely many iterations when the absolute difference between the current feasible objective function value and the maximum objective function value of the upper bounds of the polyblock is less than ϵ.

The worst case complexity of the binary linear optimization scheme is $O(2^N)$, where N is the number of channels. But the number of channels to consider would be less or equal to N since we only need to consider the channels where the QoS constraint of the SU pair is satisfied. Therefore the proposed scheme is suitable for systems with $20 - 30$ resource blocks or channels.

The polyblock based scheme to find the optimal power allocation performs multiple operations in one iteration. These operations include the following:

1. Finding the upper corner z of the hyper-box b_{max} that maximizes the objective function

2. Finding the point x where the line between the lower corner a of the hyper-box and the upper corner z of the hyperbox intersects the feasible region boundary

3. Add the new set of hyperboxes replacing b_{max} depending on x and z

4. Remove the hyperboxes which are included in larger hyper-boxes which constitutes the polyblock

5. Reduce the size of the hyperboxes so that the parts not containing a solution is not considered

As one can see the number of hyper-boxes grow by N each iteration. In order to avoid the storage problem of all the hyper-boxes generated, we only keep L boxes which gives the highest objective function values when the objective function is evaluated with the upper corner coordinates of the hyper-boxes. Therefore, at any given time, there can be a maximum of L hyper-boxes. If we use the heapsort algorithm to carry out the first operation, evaluation of the objective function has time complexity $O(LN)$ and sorting has the time complexity $O(L\log(L))$. Therefore the resultant time complexity of the first operation is $O(LN)$ assuming $L > N$. The second operation is a linear programming problem. If we use the primal-dual interior point method to solve this problem, the time complexity of this step to produce an ϵ-accurate solution is $O(\sqrt{m+n}\log(\frac{1}{\epsilon}))$ [96]. Here, m is the number of constraints and n is the number of variables. In our case, the number of variables is N and the number of constraints is $3N + 2$. Therefore, in our case the time complexity is $O(\sqrt{N}\log(\frac{1}{\epsilon}))$. The creation of new hyper-boxes have a time complexity of $O(N^2)$. Checking for the hyper-boxes which are included in another hyper-box has a time complexity of $O(L^2N)$. In order to reduce the size of hyper-boxes, for each box we need to solve two linear optimization problems for each dimension. Since we have an N dimensional problem and there are a maximum of L hyper-boxes, we need to solve $2NL$ linear single dimension optimizations problems. Therefore the complexity of it is $O(LN^{1.5}\log(\frac{1}{\epsilon}))$. When we consider all of these operations, in the asymptotic complexity analysis the operation with the highest complexity dominates. Therefore the complexity of the monotonic optimization algorithm in one iteration is $O(L^2N)$.

4.5 Estimation of Interference

In order to solve the above mentioned optimization problems we needed information on the level of current interference J_{im} on each channel i and MS m. In the above sections, we used the interference level at the MSs, at the end of the previous time slot to be the current interference. Furthermore, to ensure the available interference gap is shared among all the active users, each pair generated a random number δ. This δ depended on the number of active CR user pairs. In the method proposed here, estimation of the interference at each monitoring station is done based on the past data sent by the MSs and, the current and past data gathered at each SU pair on the observed CR interference . Since the estimated interference depended on the observed interference, sharing of the interference gap based on the random number δ was no longer needed. Because of this, the system did not have to check for the active number of CR pairs each time slot. This reduces the control overhead of sending keep-alive messages from the CRs.

Here, we assumed that the MSs as well as the SUs can distinguish between the PU interference and the secondary user interference, where as for the earlier scheme only MS had to have this capability. Then, at the end of each time slot the MSs send the SU interference observed at MSs to the base-station. In addition to that, the SUs also maintain a record of the SU interference observed at each time slot. Then, using a time series of interference observations at the MSs and at the SU pair, we found a matrix to map the interference observed at the MSs to that at the SU pair using least squares. Then, using this mapping matrix, the error covariance and the covariance of interference observed at the MSs, we can develop a Kalman filter to predict the state of the system. The state in this context is the interference observed at each MS at the time the transmission decision is taken. We will elaborate on this procedure in the next two subsections.

4.5.1 Least Squares Mapping

As mentioned before, the MSs send the data on interference $\hat{\mathbf{J}}_k(n-1)$, where $k = \{1, ..., M\}$, at the end of each time slot. Then, using the information on the power level used on the channel at time $n-1$, and the channel gains between the SU pairs and the MSs at time slot $n-1$, we calculated the interference caused by all other pairs at the MSs, $\mathbf{J}(n-1)$, as shown in Equation (4.5.1). Here, we have dropped the channel index i, since this procedure has to be run on each channel separately.

$$\mathbf{J}(n-1) = \begin{bmatrix} \hat{\mathbf{J}}_1(n-1) \\ \vdots \\ \hat{\mathbf{J}}_M(n-1) \end{bmatrix} - \begin{bmatrix} g_1^1 & g_1^2 \\ \vdots & \\ g_M^1 & g_M^2 \end{bmatrix} \begin{bmatrix} p^1 \\ p^2 \end{bmatrix} \quad . \tag{4.5.1}$$

In Equation (4.5.1), g_k^θ is the channel gain between the MS k and user θ of the pair, and p^θ is the power allocated on the channel by user θ of the pair. Let the interference from other SUs observed at the MSs, $\mathbf{J}(n)$, in the past W time slots as observed at time slot n be represented by $\mathbf{D}_{W\times M}(n)$, and the interference observed at the SU pair be represented by $\mathbf{F}_{W\times 2}(n)$. Expressions for $\mathbf{D}_{W\times M}(n)$ and $\mathbf{F}_{W\times 2}(n)$ are shown in equations (4.5.2) and (4.5.3) respectively, where $\mathbf{z}(n)$ is the observed SU interference at the pair of SUs at time n. Then, we assumed that $\mathbf{F}(n)$ can be represented as a linear combination of $\mathbf{D}(n)$ and an error term, $\kappa(n)$, as shown in Equation (4.5.4), where β is the coefficient matrix to be determined.

$$\mathbf{D}(n) = \begin{bmatrix} \mathbf{J}(n-1)^T \\ \mathbf{J}(n-2)^T \\ \vdots \\ \mathbf{J}(n-W)^T \end{bmatrix} \tag{4.5.2}$$

$$\mathbf{F}(n) = \begin{bmatrix} \mathbf{z}(n-1)^T \\ \mathbf{z}(n-2)^T \\ \vdots \\ \mathbf{z}(n-W)^T \end{bmatrix} \tag{4.5.3}$$

$$\mathbf{F}(n) = \mathbf{D}(n)\beta(n) + \kappa(n) \tag{4.5.4}$$

The least square solution for β such that the error term, $\kappa^T\kappa$, is minimized is calculated by the expression, $(\mathbf{D}^T\mathbf{D})^{-1}\mathbf{D}^T\mathbf{F}$.

4.5.2 Kalman Filtering

Kalman filtering has been proposed to reduce the errors in measurements using a combination of past trend based predicted outcome and the measured outcome. In the proposed method, we try to predict the interferences at monitoring stations using the interference observed at SUs. We focus on a single channel, since we calculate the interference of different channels independent of each other. In the least square mapping, we found the mapping matrix β, and the measurement error distribution. We use those parameters in defining the linear stochastic difference equations as in [97], which defines the system to which we applied Kalman filtering. Thus, we assume that the system in question is or can be closely approximated by a linear system. The state vector to be determined is the interference caused by all other pairs on a given channel on all the monitoring stations. Since there were M monitoring stations, we have an $M \times 1$ vector, $\mathbf{J}(n)$, as our state at time n. We assumed the current interference vector, $\mathbf{J}(n)$, can be represented as a function of the previous interference vector, $\mathbf{J}(n-1)$, and the process noise $\mathbf{w}(n-1)$, as shown in Equation (4.5.5). This equation is the first linear stochastic difference equation defining our system.

$$\mathbf{J}(n) = \mathbf{J}(n-1) + \mathbf{w}(n) \quad . \tag{4.5.5}$$

The same approach was taken in [98] to model the interference caused by other cells in a centralized time division multiple access (TDMA) packet network. If we assume we know the power allocation of all the other SU pairs, $\mathbf{P}(n-1)$, and the channel gains from them to the MSs at time slot $n-1$, $\mathbf{H}_{SM}(n-1)$, and the change of power and channel gain from the previous time slot to the current, ϵ_P and ϵ_H, we can easily see the relationship shown in Equation (4.5.6) to be true.

$$
\begin{aligned}
\mathbf{J}(n) &= (\mathbf{H}_{SM}(n-1)+\epsilon_H)\cdot(\mathbf{P}(n-1)+\epsilon_P) \\
&= \mathbf{J}(n-1)+\epsilon_H\mathbf{P}(n-1)+\mathbf{H}_{SM}(n-1)\epsilon_P+\epsilon_H\epsilon_P \\
&= \mathbf{J}(n-1)+\mathbf{w}(n) \qquad (4.5.6)
\end{aligned}
$$

In the actual scenario we do not have information about; $\mathbf{H}_{SM}(n-1)$, $\mathbf{P}(n-1)$, ϵ_P and ϵ_H. Therefore, we treated the terms having those variables as noise $\mathbf{w}(n)$. Since the noise term $\mathbf{w}(n)$ consists of multiplications and sums of independent random variables, the distribution of it can be approximated by the Gaussian distribution as the number of SU pairs increase, as per the central limit theorem. Here we assume that the time slot length is longer than the channel covariance time. Therefore, there is no correlation between the channel gains of two consecutive time slots. Furthermore, since the power level depends on the random fraction generated and the channel gains, we can assume the correlation of the power levels at two consecutive time slots to be zero or near zero. We got the cumulative distribution function (CDF) shown in Figure 4.5.1 for the case of 3 SU pairs. We also plotted a CDF of a Gaussian distribution with the same mean and variance as the error data of the experimental process, in the same figure.

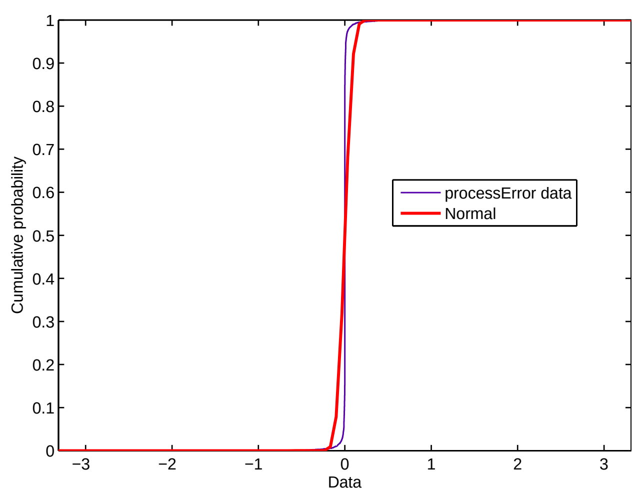

Figure 4.5.1: Normal fit for the Process error data

Equation (4.5.8) is the second equation defining our system. We assume $\mathbf{v}(n)$ to be distributed according to the Gaussian distribution with a mean of $\mathbf{0}$ and a covariance matrix $\mathbf{R}(n)$. Covariance matrix, $\mathbf{R}(n)$, for a given time slot n is calculated using the data windows $\mathbf{D}$, $\mathbf{F}$ and the coefficient matrix β as shown in Equation (4.5.7).

$$\mathbf{R} = \begin{bmatrix} \sigma_1^2 & \sigma_{1,2}^2 \\ \sigma_{1,2}^2 & \sigma_2^2 \end{bmatrix} \quad , \tag{4.5.7}$$

In Equation (4.5.7), σ_θ^2 is the variance of the entries in the θ^{th} column of κ and $\sigma_{1,2}^2$ is the covariance between the entries in the 1^{st} and 2^{nd} columns of κ.

$$\mathbf{z}(n) = \beta(n)^T \mathbf{J}(n) + \mathbf{v}(n) \quad . \tag{4.5.8}$$

In this case, the interference vector at time n can be written as a function of the power allocated by other pairs, $\mathbf{P}(n)$ as shown in Equation (4.5.9).

$$\mathbf{J}(n) = \mathbf{H}_{SM}(n) \cdot \mathbf{P}(n) \tag{4.5.9}$$

Depending on the number of MSs and the number of SU pairs, this system of equations can be solved to obtain $\mathbf{P}(n)$, assuming that $\mathbf{H}_{SM}(n)$ and $\mathbf{J}(n)$ are known (these values are not known in the actual scenario). If the number of MSs were larger than the number of SUs a least square error solution can be found. In the case the number of MSs are smaller than the number of SUs, there are an infinite number of solutions. In this case, a linear solution exists for the minimum solution, where the minimum solution $\mathbf{P}(n) = \mathbf{H}_{SM}(n)^T \left(\mathbf{H}_{SM}(n)\mathbf{H}_{SM}(n)^T\right)^{-1} \mathbf{J}(n)$. Then, the observation vector $\mathbf{z}\hat{(n)}$ can be written as shown in Equation (4.5.10), where $\mathbf{G}_i$ is a matrix of size $2 \times 2(S-1)$, having the channel gains from other SUs to SUs in pair i.

$$\begin{aligned} \mathbf{z}\hat{(n)} &= \mathbf{G}_i \mathbf{P}(n) \\ &= \mathbf{G}_i \mathbf{H}_{SM}(n)^T \left(\mathbf{H}_{SM}(n)\mathbf{H}_{SM}(n)^T\right)^{-1} \mathbf{J}(n) \end{aligned} \tag{4.5.10}$$

But it was not possible to know if $\mathbf{G}_i\mathbf{H}_{SM}(n)^T\left(\mathbf{H}_{SM}(n)\mathbf{H}_{SM}(n)^T\right)^{-1} = \beta(n)^T$ in this work. Therefore we added an error term, $\mathbf{v}(n)$, in Equation (4.5.8) to compensate for that and the thermal noise. There is no theoretical evidence to suggest that this error is Gaussian. The CDF of the experimental measurement error for the case having 3 SU pairs is shown in Figure 4.5.2. We also plot a Gaussian fit for the data alongside the experimental CDF. As one can see the measurement error has less spread than the Gaussian distribution.

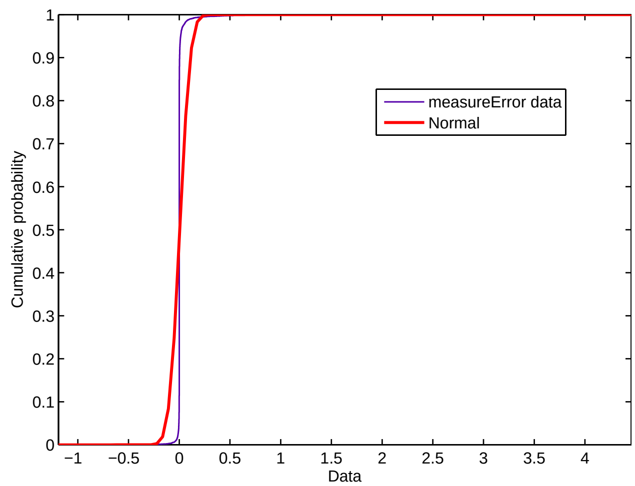

Figure 4.5.2: CDF of the Measurement error

In order to use Kalman filtering, we need to know the covariance matrix of the process noise, $\mathbf{w}(n)$. We used the past interference information in the matrix $\mathbf{D}(n)$ in calculating the covariance matrix of the process noise, $\mathbf{Q}(n)$. This process noise is also assumed to be distributed according to $\mathcal{N}(0, \mathbf{Q}(n))$.

Kalman filtering consists of two stages; the first is the time update and the second is the measurement update. The algorithm is explained in [97]. We present it here for the convenience of the reader. Let $\tilde{\mathbf{J}}^-(n)$ be the *a priori* estimate of the interference vector $\mathbf{J}(n)$ and let $\tilde{\mathbf{J}}(n)$ be the *a posteriori* estimate of the same. Then, the *a posteriori* and *a priori* estimate error covariances at time n are given by $\mathbf{S}(n)$ and $\mathbf{S}^-(n)$ respectively. As we know the exact interference vector $\mathbf{J}(n-1)$ at time n, we used this exact value as the *a posteriori* estimate $\tilde{\mathbf{J}}(n)$ when filtering the outcome for time n. Therefore, $\mathbf{S}(n-i) = 0 \quad \forall i \in \{1, ..., W\}$. This work can also be extended to the case where the MSs do not send the value of the interference vector at the end of each time slot. The time update equations are given in Equation (4.5.11).

$$\begin{aligned}\tilde{\mathbf{J}}^{-}(n) &= \tilde{\mathbf{J}}(n-1)\\ \mathbf{S}^{-}(n) &= \mathbf{S}(n-1)+\mathbf{Q}(n)\end{aligned} \tag{4.5.11}$$

The measurement update equations of the Kalman filter are shown in Equation (4.5.12), where $\mathcal{I}$ is the identity matrix of appropriate dimension.

$$\begin{aligned}\mathbf{K}(n) &= \mathbf{S}^{-}(n)\beta(n)\left(\beta(n)^T\mathbf{S}^{-}(n)\beta(n)+\mathbf{R}(n)\right)^{-1}\\ \tilde{\mathbf{J}}(n) &= \tilde{\mathbf{J}}^{-}(n)+\mathbf{K}(n)\left(\mathbf{z}(n)-\beta(n)^T\tilde{\mathbf{J}}^{-}(n)\right)\\ \mathbf{S}(n) &= \left(\mathcal{I}-\mathbf{K}(n)\beta(n)^T\right)\mathbf{S}^{-}(n)\end{aligned} \tag{4.5.12}$$

This $\tilde{\mathbf{J}}(n)$ is used as the predicted interference at the start of time slot n.

4.6 Numerical and Simulation Results

We carried out the simulations in three stages. In the first stage we compared the centralized MINLP solution with the decentralized solution based on alternative optimization. In the second stage, we compared the schemes which used interference estimation with the scheme which did not use interference estimation. Here we assumed that the past time slot data on the interference is known to the SU pair. Then, we plot the results for the cases where the interference data is available after every 1 slot, 5 slots and 10 slots respectively. In the simulations, we assumed that the PU base station calculates the required power to maintain a Signal to interference plus noise ratio (SINR) of $7dB$ assuming that the CR interference will be equal to the threshold 0.01.

4.6.1 Comparison Between the Centralized Solution and Distributed Solution

In this subsection, we compare the centralized solution for a case having 2 pairs of CRs ($M = 2$), 2 PUs ($N = 2$) and 4 MSs ($R = 4$) with the distributed solution without estimation. To get the centralized solution, we solved optimization problem (4.1.1) using OPTI [99] Matlab toolbox with the NOMAD solver [100]. The results for the mean throughput of a CR pair, the fraction of time the PU SINR is below their intended SINR (outage), the mean SINR of PUs and the fraction of time the interference condition on PU receiver is violated are plotted in figures 4.6.2, 4.6.3, 4.6.4 and 4.6.5 respectively. These results were obtained simulating the network for 1000 time slots for the network setup shown in Figure 4.6.1.

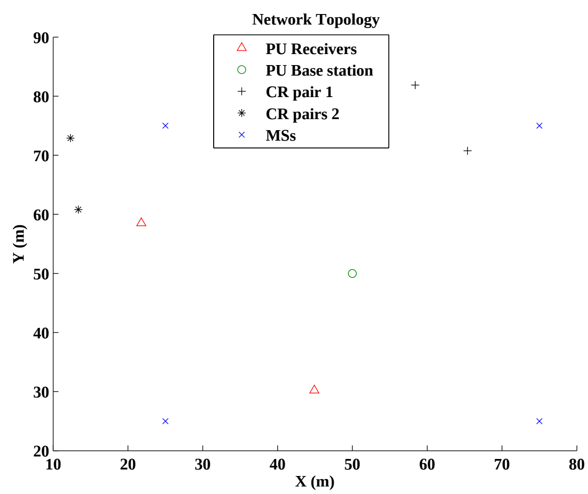

Figure 4.6.1: Network setup

In Figure 4.6.2, we can see that the data rate of the decentralized scheme is approximately twice of that of the centralized scheme, but the interference caused by the decentralized scheme is also twice as much, as shown in figures 4.6.3 and 4.6.5. The reason for this behavior is, not exchanging channel selection and power allocation information between the D2D user pairs and the usage of previous time slot information on the level of interference caused by the other D2D users at the MSs in our proposed scheme. The decentralized solution reduces the computation effort exponentially while keeping the control overhead

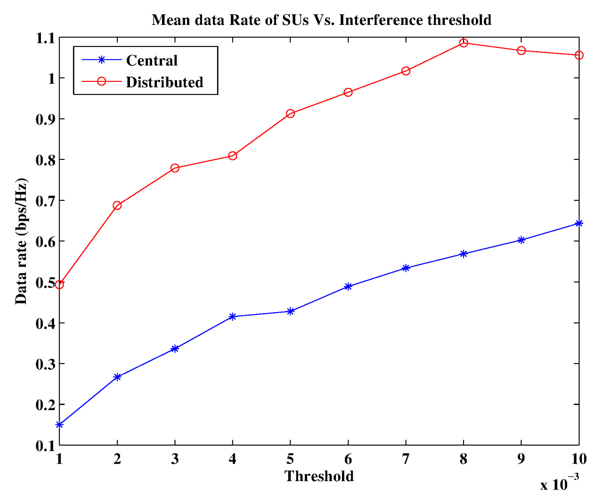

Figure 4.6.2: Mean data rate of a CR pair

minimum. In the centralized solution the channel gains between the user pairs and between the user pairs and the MSs should be sent to the central entity. In the proposed algorithm only the interference information should be sent from the base station. This reduces the control overhead significantly.

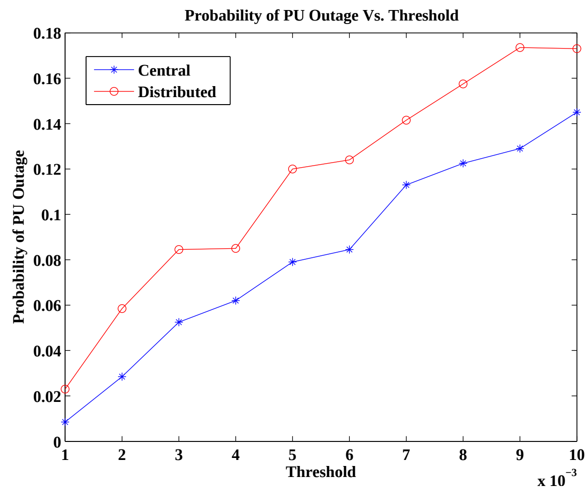

Figure 4.6.3: Probability that PU SINR is less than $3dB$

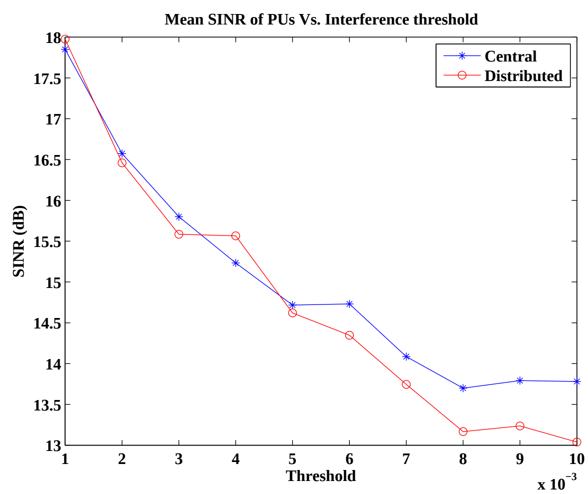

Figure 4.6.4: Average PU SINR

4.6.2 Case Where the Interference Data on the Previous Time Slot is Available

In the numerical calculations carried out in this section, we assume the random network topology shown in Figure 4.6.6. Here, the primary user (PU) base station is located at the center of the $100m \times 100m$ grid. A comparison with [49] was not possible since the coefficients of the proportional, integral and derivative parts were not given in that paper.

In the simulations, we considered Rayleigh fading and log-normal shadowing together with a path-loss model having a path-loss exponent 2.7. The standard deviation of the log-normal distribution used was $11.8dB$. We tested this scheme with 3, 5 and 8 CR pairs which

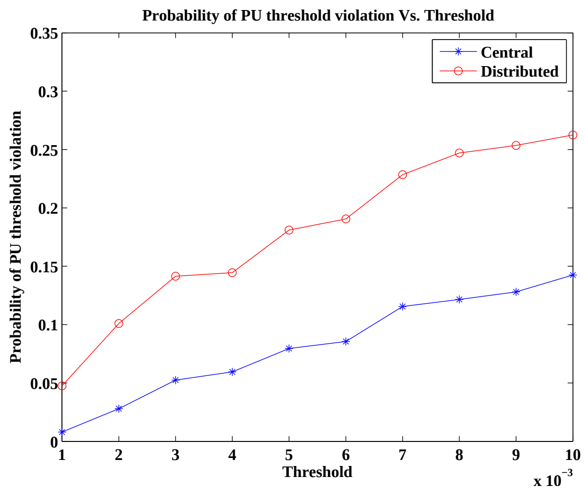

Figure 4.6.5: Probability that PU interference threshold is violated

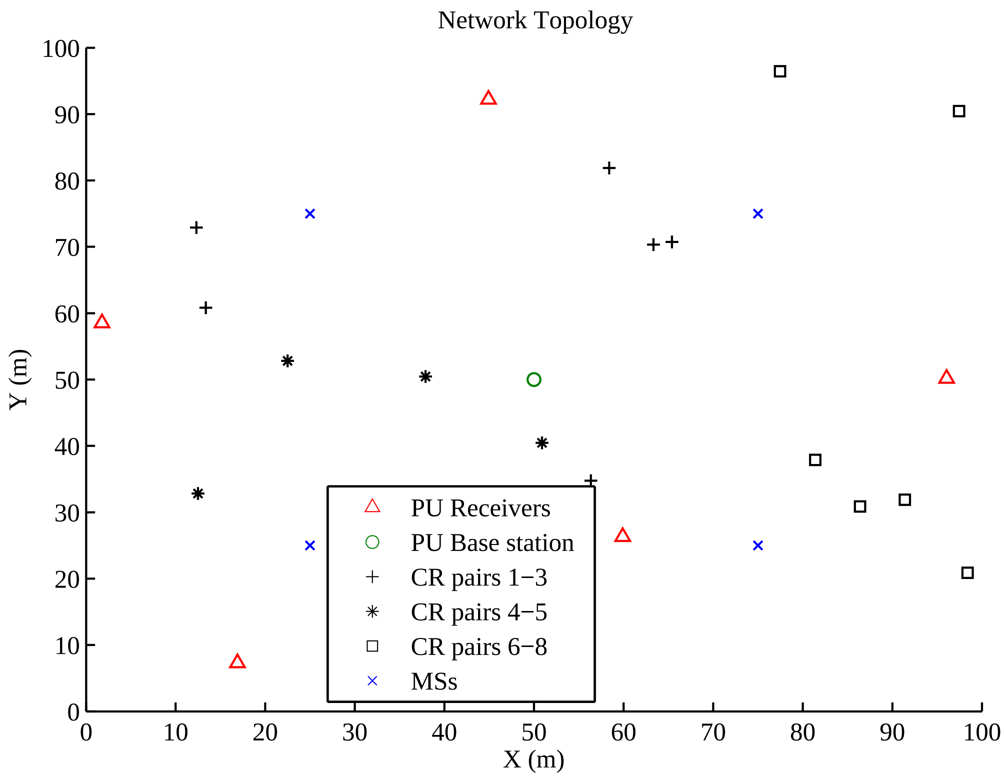

Figure 4.6.6: Multiple CR pair network setup

utilized 5 channels used by the 5 PUs. Using this network setting, we simulated the network for 10000 time slots and plotted the mean throughput of a CR pair, the fraction of time the PU SINR is below their intended SINR (outage), the mean SINR of PUs and the fraction of time the interference condition on PU receiver is violated in figures 4.6.7, 4.6.8, 4.6.9

and 4.6.10 respectively. In these figures, the plots for the interference estimation scheme are labeled with the title *With Prediction* and the ones which does not use interference estimation are labeled with *W\Prediction*.

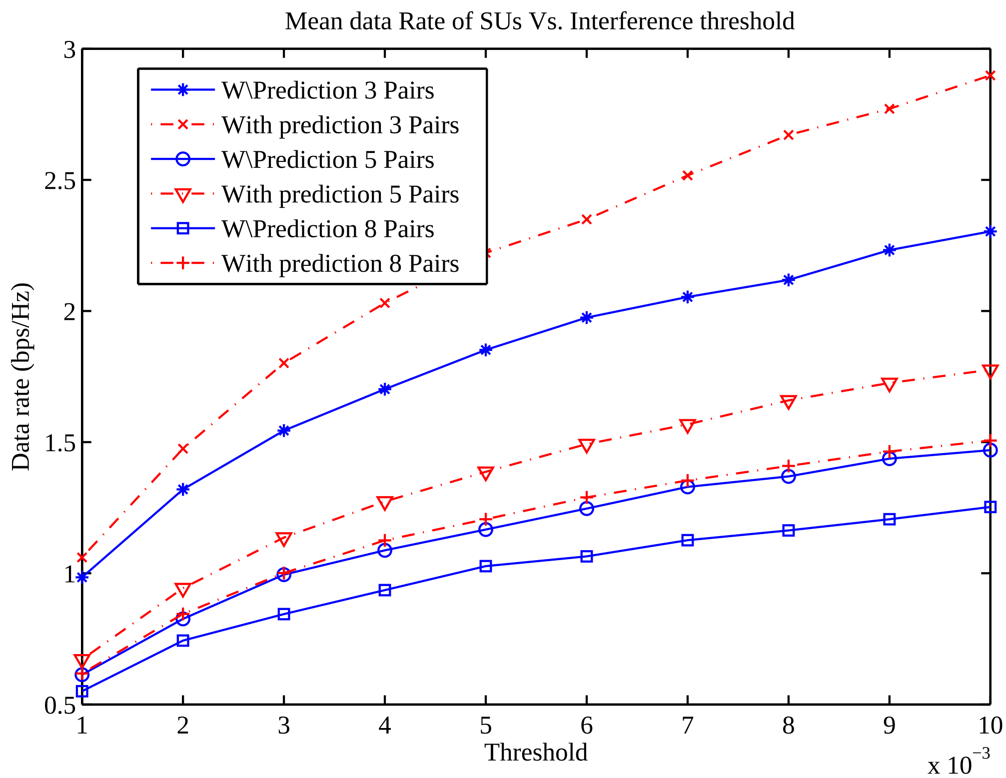

Figure 4.6.7: Mean data rate of a CR pair for different M

In Figure 4.6.7, one can see that the scheme with interference estimation outperforms the scheme without estimation, but it has a higher probability of outage for the PUs, as shown in Figure 4.6.8. In this context, a scenario where the PU SINR is less than $3dB$ is considered an outage. If we compare the two schemes at similar outage probabilities, the data rate is almost the same. In the scheme without estimation, the outage probability stays almost the same with increasing number of SUs at low thresholds. This can be a result of sharing the interference gap based on the number of active users.

When we compare the mean SINRs shown in Figure 4.6.9 and the probability of threshold violation shown in Figure 4.6.10 for the two methods, we can see that the scheme with estimation causes more interference to the PUs. This is because the interference estimation scheme under estimates the interference.

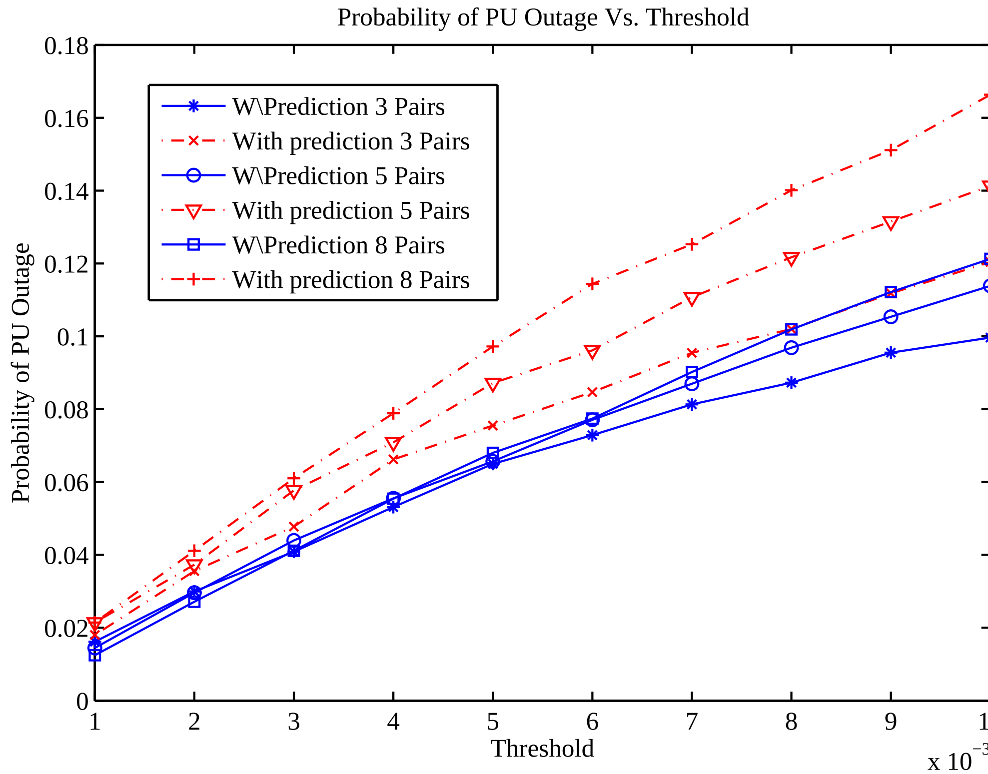

Figure 4.6.8: Probability that PU SNR is less than $3dB$ for different M

4.6.3 Case Where the Interference Data on the Previous Time Slot is Unavailable

Here, we plot the same results for the case where the interference data on the previous time slot is not available. The results we got for the cases where the interference data were available after every 1, 5 and 10 time slots are shown in figures 4.6.11, 4.6.12, 4.6.13 and 4.6.14 respectively. Let us symbolize this time lag in the interference data by τ.

When we compare the scheme with estimation to the scheme without estimation in the case where the actual interference information is available after every 5 slots and 10 slots, we could see that the data rate of the scheme with estimation is approximately half of the scheme without estimation. On the other hand, the probability of outage of the scheme with estimation is lower than the scheme without estimation for the threshold range from $0.005 - 0.01$. But the probability of outage of the scheme without estimation is lower for the interference threshold range $0.001 - 0.004$. However, we did not check the sensitivity

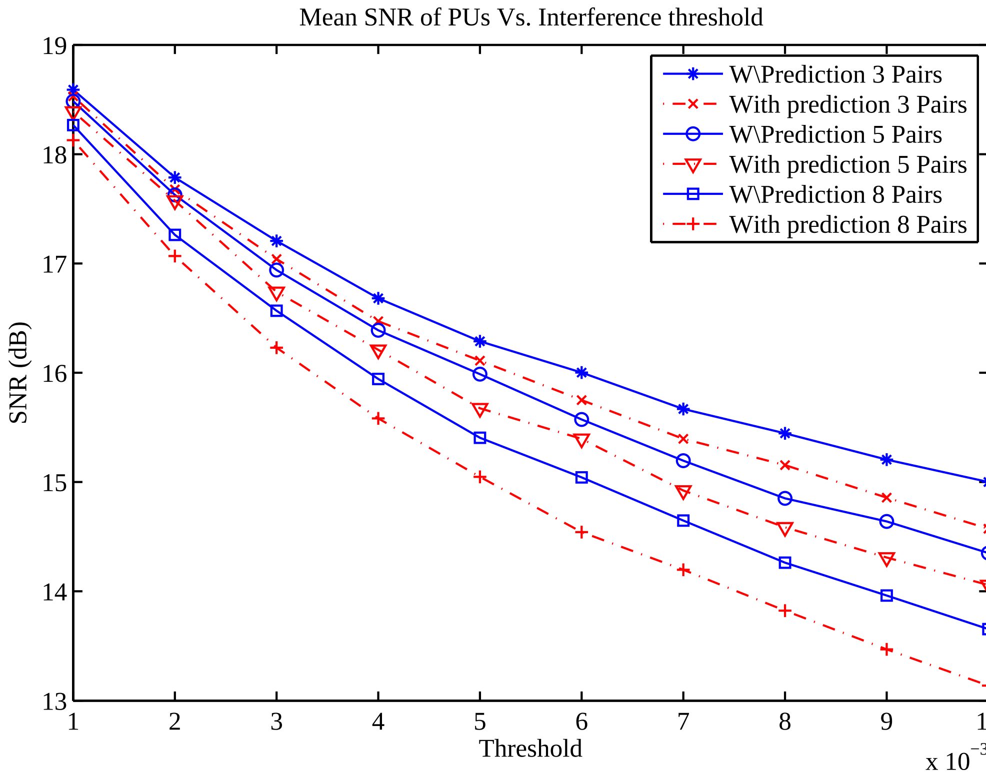

Figure 4.6.9: Mean SNR of PU for different M

of scheme without estimation on the number of active users M. When the information is sent after every τ time slots, it is possible for the actual number of active users to differ from the information the user pairs have on this number. We generate the results for this case in Section 4.6.4.

In general, when evaluating the results for the mean data rate per CR pair, one can see that the data rate reduces as the number of users increase. This was a result of the available interference gap being shared among a larger number of users.

4.6.4 Case where the Interference Data on the Previous Time Slot is Unavailable and the Number of Users are Varying

Here, we plot the results for the case where the interference data on the previous time slot is unavailable while the number of active users are varying. In this experiment, to make the number of active users vary, we selected a number between 1 and 8 with equal

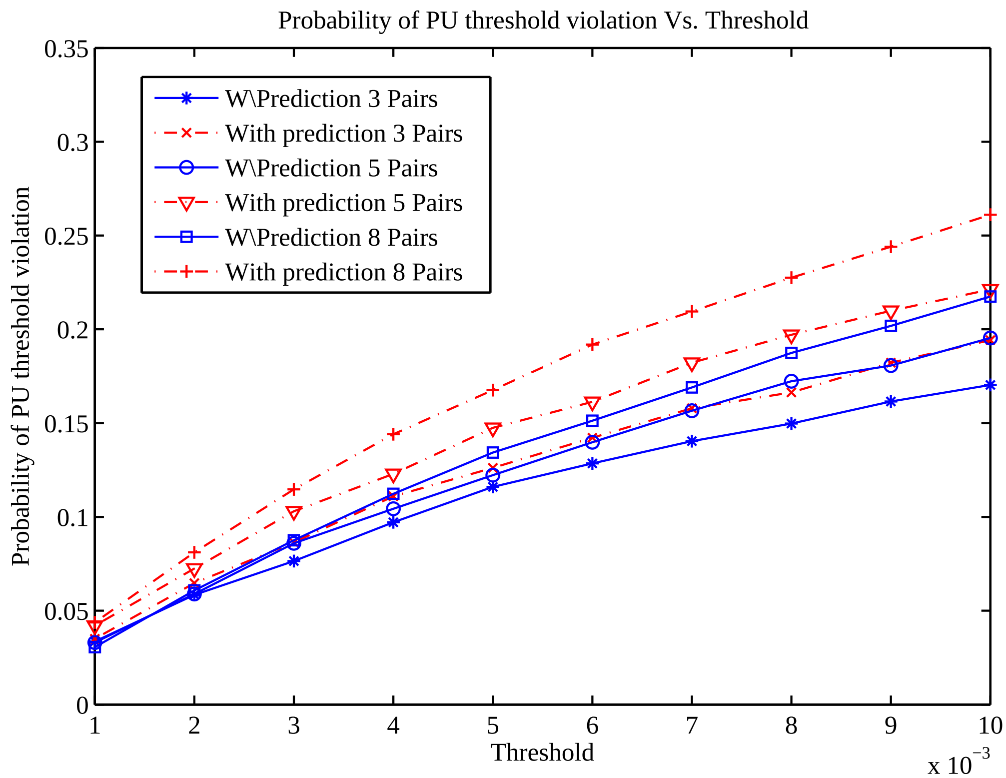

Figure 4.6.10: Probability that PU interference threshold is violated for different M

probability. This number was used as the active number of users. After this number was generated, we selected that number of user pairs randomly from the total available user pairs such that each pair had an equal probability of getting selected. In the scheme without estimation, the exact number of active users were only known when the MSs broadcast that information, since the number of active users are varying. The simulation results for this case are presented below.

For the case of varying number of active users, the scheme without estimation had a higher data rate which was approximately twice that of the scheme with estimation as shown in Figure 4.6.15. But, the probability of outage of the PUs were also higher in the scheme without estimation, as can be seen in Figure 4.6.16. Therefore, the interference threshold plays a vital role in the scheme without estimation, since the outage probability grows rapidly with the threshold.

In Figure 4.6.17, we present the mean Signal to interference plus noise ratio of the PUs. Here, we can see that the scheme with estimation has higher SINR compared to the scheme

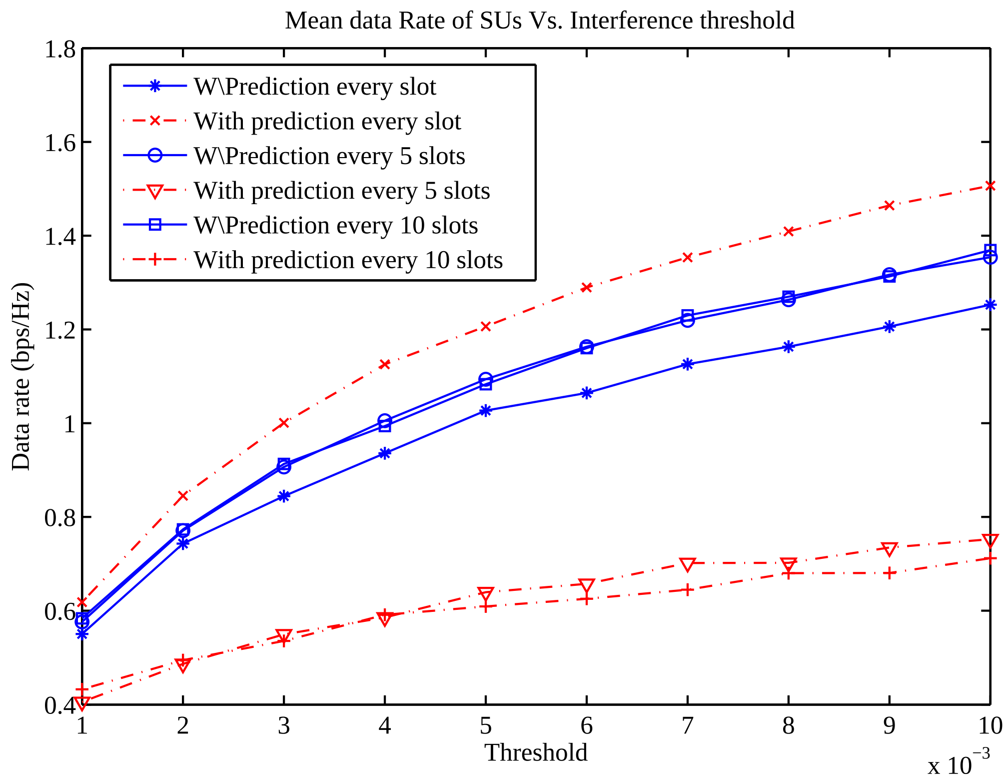

Figure 4.6.11: Mean data rate of a CR pair for different τ

without estimation. This is obviously because the scheme with estimation over estimated the interference at the MSs and transmitted with less power.

In Figure 4.6.18, we present the probability of interference threshold violation at the PU. Here also, we can see that the scheme without estimation violated the interference threshold at the PUs twice as much as the scheme with interference estimation.

4.7 Conclusion

In this chapter, we proposed a network assisted joint channel assignment and power allocation scheme which was capable of selecting if the channel should be used in the FD mode or HD mode. We decomposed the problem into two sub-problems, one dealing with the channel assignment and the other with power allocation. We solved these two problems alternatively until the change in the result is less than a threshold. We used the general branch and bound solution approach to solve the channel assignment sub-problem. Then,

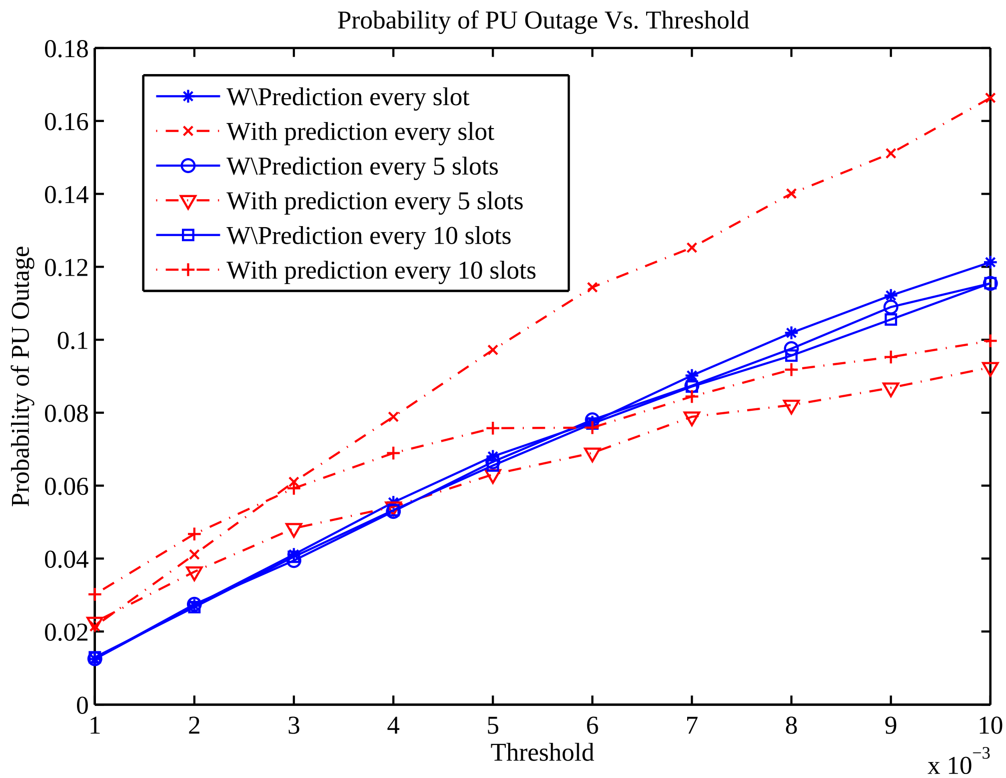

Figure 4.6.12: Probability that PU SNR is less than intended for different τ

for the power allocation sub-problem, we used the fractional programming approach which made use of the Monotonic optimization framework. Here, no assumptions about the SINR of CRs were required. Then we proposed an interference estimation scheme which does not need the information on the number of active users. In this estimation based scheme, the network is alleviated from both checking whether the user pairs are active and sending that information along with the interference information. The accuracy of the prediction algorithm can be gauged by evaluating the outage probability and the mean data rate. We need to look at both metrics, since over estimation of interference is going to keep the data rate low while keeping the probability of PU outage lower. As the simulation data suggest, the estimation scheme tend to underestimate the interference when the previous time slot interference data is available. Then it over estimates the interference when the interference information is available every 5 and 10 time slots. We consider this as the price to pay for the reduction in the control overhead. When we compared the results for a given outage probability both schemes performed almost the same in terms of data rate. Then we tested

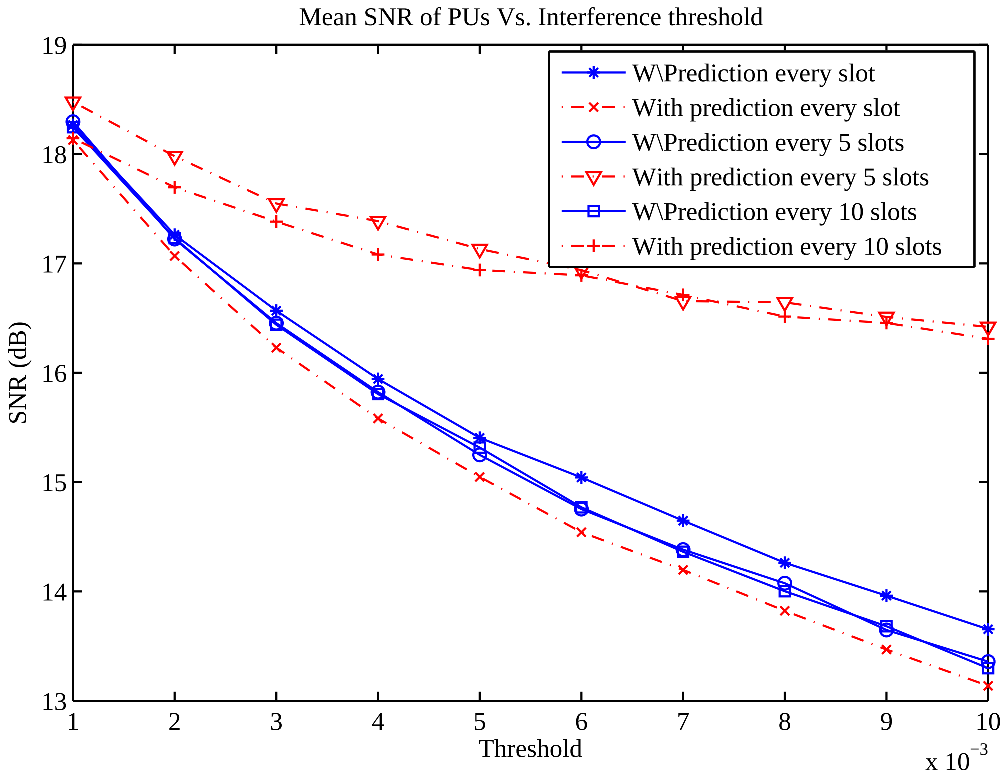

Figure 4.6.13: Mean SNR of PU for different τ

the schemes for the case where the data on interference are sent every 5 and 10 time slots having the number of SUs constant throughout the simulation. In this case, the estimation based scheme did not perform that well in terms of data rate. Then we simulated a scenario where the SUs randomly decided whether to transmit or not. In this case, the data on the interference and the number of active users were transmitted every 5 and 10 time slots. Here, the scheme without estimation did not have the information on the current number of active users. When we compared the two schemes we found out that the scheme without estimation still had a better data rate, but the interference it caused was also high. So when considering the burden of control overhead and the level of interference incurred by the scheme without estimation, we believe that the scheme with estimation is the better choice.

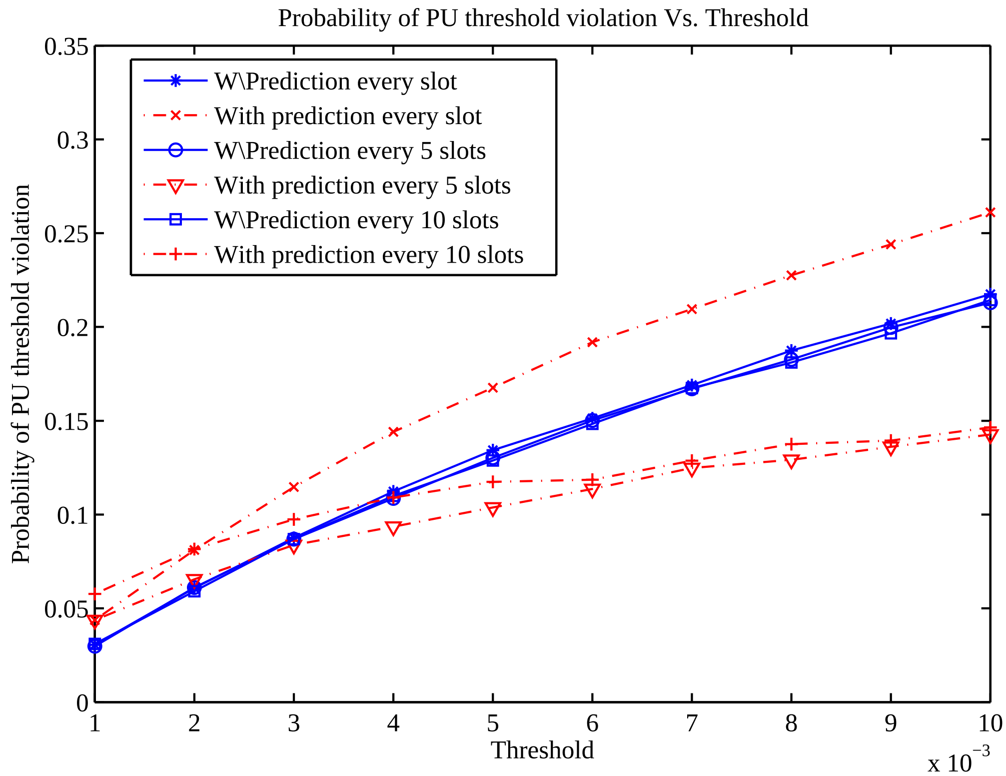

Figure 4.6.14: Probability that PU interference threshold is violated for different τ

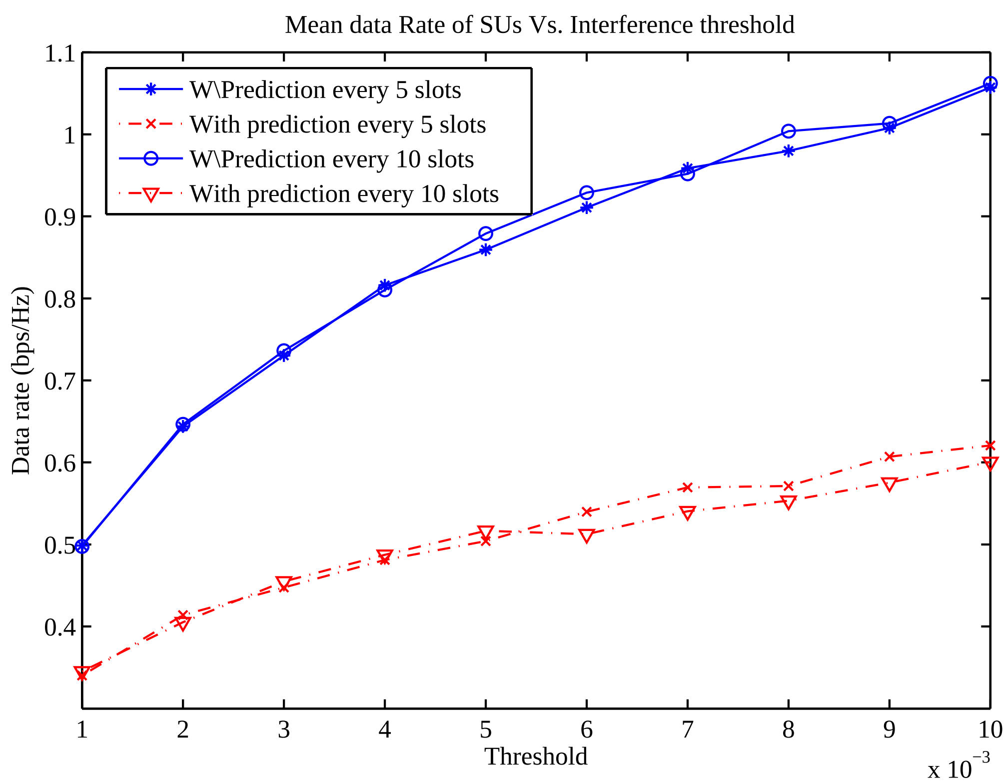

Figure 4.6.15: Mean data rate of a CR pair for different τ and varying number of Active users

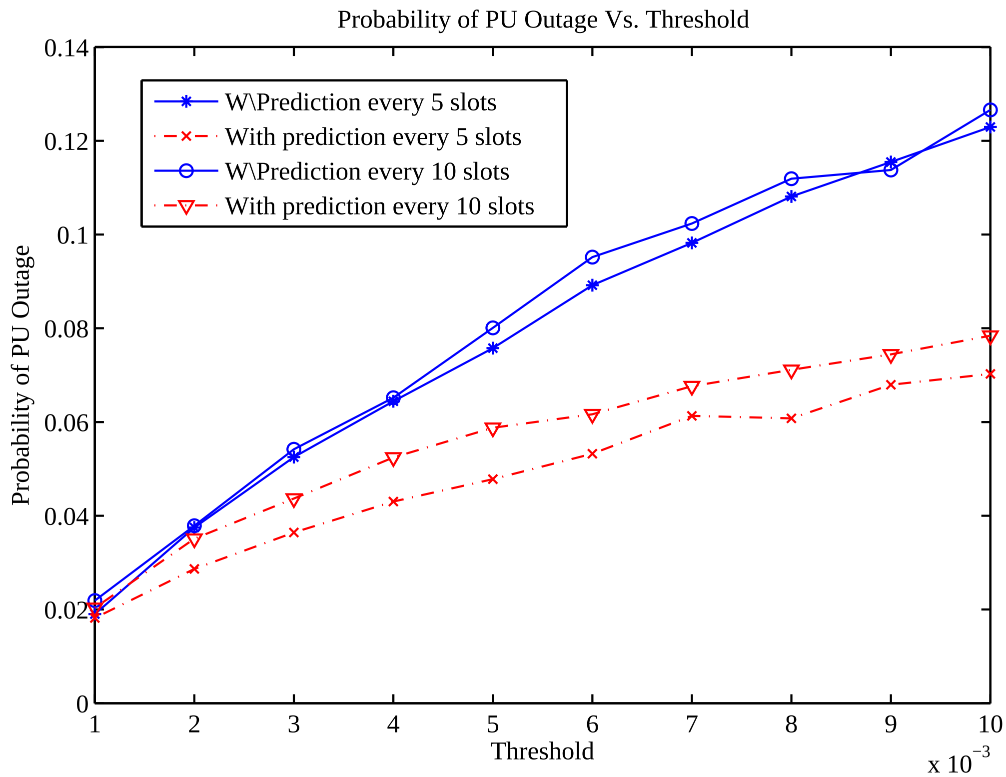

Figure 4.6.16: Probability that PU SNR is less than intended for different τ and varying number of Active users

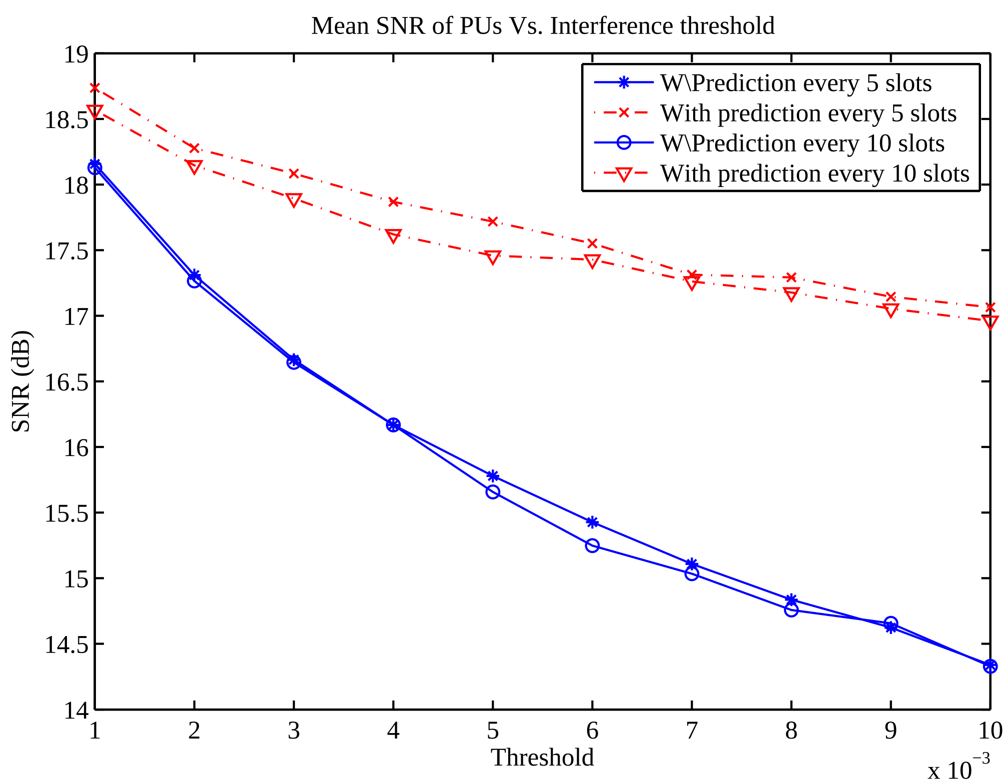

Figure 4.6.17: Mean SINR of PU for different τ and varying number of Active users

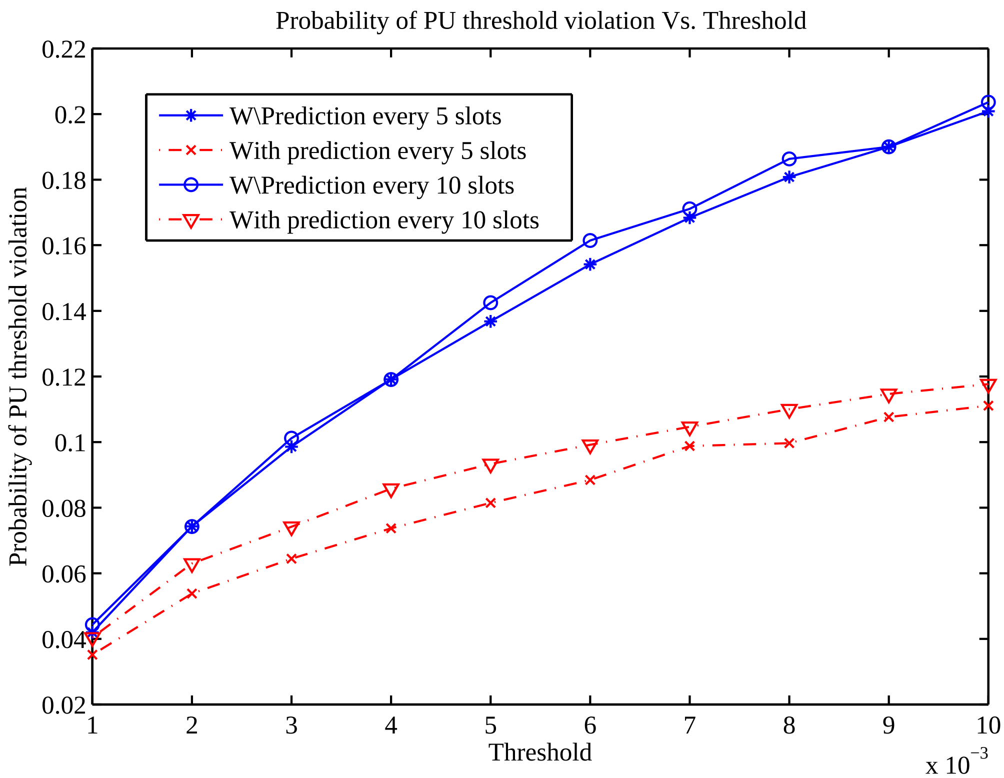

Figure 4.6.18: Probability that PU interference threshold is violated for different τ and varying number of Active users

Chapter 5

Conclusion and Future Extensions

In this chapter, we present the concluding remarks that can be drawn from the experiments we carried out based on the model, and then we discuss several possible foreseeable extensions to the work presented in chapters 3 and 4.

5.1 Concluding Remarks

In this thesis we showed the importance of estimation and prediction techniques in the cognitive radio scenario both in the overlay type half-duplex networks and in underlay type full-duplex networks. Although many papers applying machine learning, Markov decision process and probability theory on overlay networks are available in the literature, not much work are there applying estimation techniques on underlay networks. There are some work in the literature on the overlay FD networks, but they focus only on the ability of the CRs to sense while transmitting. A significant number of research papers addressing the resource allocation problem of half-duplex CR networks are present in the literature, but none of these schemes used interference estimation. Our interference estimation scheme can be utilized in these schemes too, given that they follow our network topology.

In Chapter 3, we presented a proactive medium access control scheme for an overlay cognitive radio network. There we showed the importance of using both the primary user channel usage statistics as well as secondary user channel usage statistics in selecting a channel set for transmission. We showed that using the secondary user statistics, the probability of successful transmission can be increased compared to selecting a channel set at random. Obviously, if the secondary users only used the primary user statistics in selecting the channel set, each user will have the same channel set arrangement. Therefore, the number of collisions between the SUs brings the probability of successful transmissions down, and the secondary users will have to idle for a long time to find a channel that can be used. In this scheme, first we proposed a sequence hopping common control channel and derived the theoretical properties of it. Here, selecting the sequence of channels was done based on the channel utilization probabilities of the primary users. Therefore the secondary users used the channels with low primary user utilization with higher probability. Then we proposed two methods to select the channels for the data transmission. In the first method, we found the set of channels based on the product of the probability of the channel being idle and the probability of winning the contention on that channel. In the second method, we used a marked Markov process based technique to find the net reward each channel produces for the next Q number of time slots. Then, the set of channels having the highest net reward was chosen as the set of channels to be used in the transmission. In the simulations, we showed that our schemes provided better probability of successful transmission while producing insignificant interference to the primary users.

In Chapter 4, first we presented a joint channel assignment and power allocation scheme for a full-duplex underlay cognitive radio network which used a random fraction of the interference gap present. Here, we assumed each user pair decides on the channels and the level of power to be used based on the interference gap present at the monitoring stations. Based on the channel selection of the pair of secondary users, the transmission mode selected for a given channel can be either half-duplex or full-duplex for a given channel. In order to solve this joint problem, we decomposed it into two sub-problems one dealing with channel assignment and the other dealing with power allocation. Then we alternatively solved those till there is no significant change in the allocated power. The channel assignment problem was a linear binary optimization problem, which we solved using the general branch and bound approach. The power allocation problem was solved using the monotonic optimization framework. This solution method depended on the number of secondary user pairs in the network. Then, considering the control overhead of finding out the number of active secondary users in each time slot, we proposed a scheme which estimated the interference at the monitoring stations based on the local interference. In this scheme, we used least squares mapping to find the parameters which defined the Kalman filter. In the simulations, we showed that the estimation based scheme performed very well in the different network setups we used to test the scheme.

As the conclusion, we can say that resource allocation in cognitive radio networks can take advantage of the estimation and prediction techniques to improve the data rate for the secondary users while minimizing the interference to the primary users in both overlay networks and underlay networks.

5.2 Future Directions

5.2.1 Proactive Channel Access in Full-duplex Overlay Networks

In Section 2.3.2 we discussed some schemes on full-duplex cognitive radio. In these schemes discussed in Section 2.3.2, the authors did not consider the requirement for the secondary users to find an alternative channel in the event that the current channel gets occupied. In [72] and [74], the authors used the ability of the full-duplex cognitive radios to transmit and sense at the same time to sense the channel while transmitting. Thereby, they aimed to minimize the time period the primary user transmission overlaps with the secondary. In these schemes the channels were chosen at random and were sensed continuously while the transmission was going on. If the primary user activity distributions are known, it is possible to design a channel sensing pattern to minimize the interference to primary users while having knowledge of the status of other channels. This knowledge of the state of other channels could be used to switch the channel in the event the primary

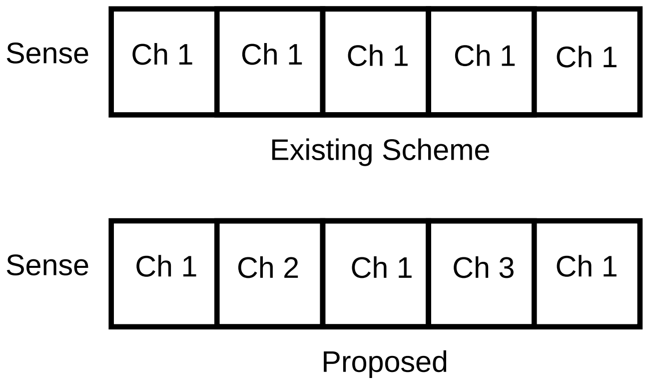

Figure 5.2.1: An example proactive FD sensing scheme

user tries to access the channel. Therefore, using channels that have a lot of primary user activity is harmful in two ways. The first is we need to check them for activity more often which limits the number of other potential channels we can check. The other is secondary users transmission will get interrupted more often. An example diagram is shown in Figure 5.2.1. The top figure shows what is proposed by [72] and [74] and the one under it shows an example of what we propose. In the top figure channel 1 is continuously sensed while transmitting on the same channel. In the bottom figure channels 2 and 3 are sensed in addition to channel 1 while transmitting on channel 1.

Here, the parameters that should be found is the number of channels to sense and when that sensing should be done such that the interference caused on the primary users is below a threshold. Therefore, one should form an optimization problem to maximize the number of other channels sensed while keeping the probability of interference below a threshold. This scheduling problem is not straight forward since the probability of the primary user returning depends on the time elapsed since the primary user went from the busy state to the idle state. This can be simplified if we use a memoryless distribution like the Exponential distribution. But, this will be far from reality.

5.2.2 Interference Threshold Calculation

We assumed the interference threshold is known in the results obtained in Chapter 4. The interference threshold at each monitoring station was assumed to be the same. This assumption actually limits the special degree of freedom of the secondary users. Ideally, the monitoring station nearest to the primary receiver should have a lower threshold on the channel that particular primary user is using while the monitoring stations far away should have higher thresholds for that channel. But, calculation of these thresholds at each monitoring station should require either some knowledge of the signal to interference plus noise ratio at the primary receivers or the number of retransmissions from the base station to the primary receiver. Based on this information each base station should calculate the thresholds for each channel separately. Therefore, there should be some closed loop control in place to set the thresholds.

Appendix A

PST learning algorithm

This algorithm is based on the papers [82] and [83]. It consists of two phases. The main purpose of the algorithm was to find out a hypothesis which resembles the distribution that generated the sequence, where the Kullback-Leiber distance [101] between the actual and the empirical distribution is less than ϵ per state with probability $1-\delta$, where $0<\epsilon<1$ and $0<\delta<1$ [82]. For this algorithm to generate a good hypothesis, a training sequence of sufficient length representing the general channel usage is needed. In this algorithm, empirical probabilities of the occurrence of any string s, $\tilde{P}(s)$, and the occurrence of a particular observation σ given an observation history sequence, s, $\tilde{P}(\sigma|s)$, are used. The formulae in [82] for calculating the above mentioned probabilities are shown in equations (A.0.1) and (A.0.2) respectively. In calculating these values, we assumed that a training sequence $\sigma_1^m = \sigma_1\sigma_2\ldots\sigma_m$ of length m is provided.

$$\tilde{P}(s) = \frac{1}{m-D+1}\sum_{j=D}^{m-1}\chi_j(s), \tag{A.0.1}$$

$$\tilde{P}(\sigma|s) = \frac{\sum_{j=D}^{m-1}\chi_{j+1}(s\sigma)}{\sum_{j=D}^{m-1}\chi_j(s)}. \tag{A.0.2}$$

In equations (A.0.1) and (A.0.2), $\chi_j(s)$ is an indicator function which takes value 1 when $s = \sigma_{j-|s|+1}\ldots\sigma_j$ and 0 otherwise, where $|s|$ is the length of string s and D is the order of the model. As an example, let $\sigma_1^{10} = 1110010000$ and $s = 00$. When $j = 2$,

$\chi_2(00) = 0$ since $\sigma_1\sigma_2 = 11$ is not equal to $s = 00$. When $j = 5$, $\chi_5(00) = 1$, since $\sigma_4\sigma_5 = 00 = s$. All Markov predictors depend on the property that, the probability distribution of the next symbol can be approximated by conditioning it on the previous D symbols, which is called by the name "*short memory principle*" [102]. If we chose the memory D to be too small than required, the distribution is going to be incapable of capturing all the dependencies between the symbols, which degrades prediction efficiency [102]. On the other hand, if we chose it to be too high, it over-fits the training sequence and gives a high prediction error.

The probabilistic suffix tree algorithm has a favorable property of avoiding the over-fitting of the model to the training sequence if the maximum memory length D chosen was too high. Therefore, we can afford to keep D to be higher than required, since the algorithm takes care of it [102]. But this advantage did not come free. For the algorithm to take care of this issue, more tunable parameters were introduced in [102]. This algorithm is governed by five parameters namely D, P_{min}, α, r, and γ. These parameters control the upper bound of the number of states in the variable order Markov model. The first parameter is the maximum memory of the variable order Markov model, denoted by D. The value of D is the maximum number of ones and zeros in the strings in set S which are used to name the states of the Markov Chain or the maximum level to which the tree is grown. The second parameter is the minimum probability of abundance, P_{min}, of any binary string s' of length $|s'|(\leq D)$ in the training sequence for s' to be a member of the tree. This is a necessary condition, but it is not sufficient. The third parameter α denotes the minimum value the conditional probability, $\tilde{P}(\sigma|s')$, (see Equation (A.0.2)) can take for a binary string s' of length $|s'|(\leq D)$ and a symbol $\sigma \in \{1, 0\}$. This is also a necessary condition a string s' should satisfy in order for it to be a node on the tree. The fourth parameter r is a threshold, which determines whether the string s contributes additional information in predicting the next symbol σ than its longest suffix $\hat{s}$, which is already a node on the tree (if $s = \sigma_1\sigma_2...\sigma_{k-1}\sigma_k$ then $\hat{s} = \sigma_2...\sigma_{k-1}\sigma_k$). The final parameter γ is the probability assigned to $\tilde{P}(\sigma|s)$ if the value of it is zero for any symbol σ and state s. Phase 1 of the

PST algorithm is outlined in Algorithm 4.

Algorithm 4 PST Learning Algorithm
Phase 1

Initialize $\bar{T}$ and $\bar{S}$: $\bar{T}$ is a binary tree having a root node e denoting an empty string & $\bar{S}$ is an empty set
$\bar{S} \Leftarrow \{\sigma | \sigma \in \{1, 0\}$ and $\tilde{P}(\sigma) \geq P_{min}\}$
while $\bar{S} \neq \emptyset$, pick any $s \in \bar{S}$ and **do**
    remove s from $\bar{S}$
    if $\exists \sigma \in \{1, 0\}$ s.t $\tilde{P}(\sigma|s) \geq \alpha$ and $\frac{\tilde{P}(\sigma|s)}{\tilde{P}(\sigma|Suffix(s))} > r$ **then**
        add the nodes to $\bar{T}$ corresponding to s, which includes all the nodes on the path starting from the deepest node which is a suffix of s already in $\bar{T}$ up to s
    end if
    if $|s| < D$ **then**
        $\Sigma' = \{1, 0\}$
        while $\Sigma' \neq \emptyset$ **do**
            remove σ' from Σ'
            if $\tilde{P}(\sigma' s) \geq P_{min}$ **then**
                add $\sigma' s$ to $\bar{S}$
            end if
        end while
    end if
    Continue to next $s' \in \bar{S}$
end while

As one can see, in the above algorithm we start with a tree having a single node representing the null string e, and as the algorithm progresses we add nodes corresponding to the string suffixes. Here, we only add the suffixes which are mandatory for the distribution to be correct. We add a node v labeled by a string s to the tree, $\bar{T}$, if the following criteria are satisfied by the string s. First, the empirical probability of the occurrence of string s in the training sequence, $\tilde{P}(s)$, should be larger than a threshold P_{min}. Because of this requirement, we avoid the exponential growth of the number of strings to be tested in the algorithm. Then, we check whether the probability of occurrence of a symbol $\sigma \in \{1, 0\}$ after the string s, $\tilde{P}(\sigma|s)$, is greater than a threshold α. Thereafter, we check to see whether $\tilde{P}(\sigma|s)$ is greater than the probability of getting σ after the longest suffix $suffix(s)$ of s. We check this by taking the ratio, $\frac{\tilde{P}(\sigma|s)}{\tilde{P}(\sigma|suffix(s))}$, and finding out whether it is greater than r, where $r > 1$. Next, we add the extended version of s which is σs, where $\sigma = \{1, 0\}$, to the set of strings $\bar{S}$. We select strings which are added to the tree from the set $\bar{S}$. If the probability of occurrence of σs in the training sequence is greater than P_{min}, we add the node corresponding to σs to the tree immaterial of whether s or a suffix of s is included in the tree or not. This is done because it is possible to have a string s' who has a conditional distribution substantially different from its parent. Parent of a node in our tree is labeled by the longest suffix of that particular node.

After performing Algorithm 4 on the training sequence there is a possibility that, for some of the nodes, the probability of occurrence of either 1 or 0 after the string s labeling that node ($\gamma_s(1)$ or $\gamma_s(0)$) is zero. Further more there is a possibility that the internal nodes of the tree $\bar{T}$ may only have one of the children with respect to the occurrence of either 1 or 0. We correct those problems in the phase 2 of the algorithm given in Algorithm 5 [82].

Although we introduced this as a Markov model, the algorithm grows a tree. In this tree the leaf nodes give the respective state labels of the equivalent Markov chain. But in the case that the destination state after a transition from a given state is ambiguous we should add some additional states. Since the tree is sufficient for the predictions to be done, the readers are referred to [82] for more information about constructing the equivalent Markov chain.

Algorithm 5 PST Learning Algorithm
Phase 2

1: Initialize $\hat{T}$ to be $\bar{T}$
2: Extend $\hat{T}$ by adding all missing sons of internal nodes.
3: For each node in $\hat{T}$ labeled by string s calculate the conditional probability $\gamma_s(\sigma)$ for each $\sigma = \{1, 0\}$ using Equation:
4: $\gamma_{s'}(\sigma) = \tilde{P}(\sigma|s')(1 - 2\gamma) + \gamma$,
5: where $s' = s$ if s is in $\bar{T}$ else s' is the longest suffix of s in $\bar{T}$.

Appendix B

Monotonic optimization framework

As per the brief explanation in Chapter 4, this algorithm starts with a hypercube which covers the entire feasible region. In this algorithm Tuy et al. defined a construct called a polyblock, which is the normal hull of a vertex set $\mathcal{V}$. This vertex set $\mathcal{V}$ contained the generated upper corners of the hypercubes. This polyblock $\mathcal{Q} = \cup_{\mathbf{z}\in\mathcal{V}}[\mathbf{P}, z]$. Therefore, polyblock $\mathcal{Q}$ represents the area covered by all the hypercubes having their upper corners in $\mathcal{V}$ and the lower corner in $\mathbf{P}$. A proper polyblock is defined as, a polyblock which has vertices that are not strictly element-wise less than the other vertices in $\mathcal{V}$. In this case $\mathcal{V}$ is called a proper vertex set. Let us assume our objective function is $f(\mathbf{p})$, and the feasible set of values is given by $\mathcal{G}$. Let the proper vertex set be represented by $\mathcal{V}^k$ in the k^{th} iteration, and z^k is the vertex that maximizes $f(.)$. Then if $z^k \in \mathcal{V}^k$ and $z^k \in \mathcal{G}$ then z^k is the optimal value. Otherwise we do two operations. The first one is selecting a point x^k which is at the intersection of the Pareto boundary of the feasible set and the line from the lower corner a of a hypercube to the upper corner z^k. The second is the generation of a new set of proper vertices, $\mathcal{V}^{k+1}$, for the new polyblock $\mathcal{Q}^{k+1}$. Details of these two steps are explained in the below subsections.

Computation of x^k

For a given point z^k, a feasible set $\mathcal{G}$ and an objective function $f(.)$, we find the boundary point x^k using Equation (B.0.1), where λ^k is given in Equation (B.0.2).

$$x^k = z^k - \lambda^k \left(z^k - a\right) \quad , \tag{B.0.1}$$

$$\lambda^k = \min\left\{\lambda | \left(z^k - \lambda(z^k - a)\right) \in \mathcal{G}\right\} \tag{B.0.2}$$

Calculation of the new polyblock $\mathcal{Q}^{k+1}$

After we generate the farthest point x^k on the boundary of the feasible region, we make the polyblock smaller by shrinking its vertices. Before investigating the method of shrinking, let us carry forward the definition of function $J(z, y)$ from [92] in Equation (B.0.3).

$$J(z, y) = \{j | z_j > y_j\} \tag{B.0.3}$$

Let the improper polyblock T' be defined as in Equation (B.0.4), which is the first step in generating $\mathcal{Q}^{k+1}$, where set $\mathcal{V}_* = \{z \in \mathcal{V}^k | z > x^k\}$, $\mathbf{e}^i$ is the i^{th} column of an identity matrix of size $2N \times 2N$, and x_i^k and z_i represent the i^{th} element of the respective vectors.

$$\begin{aligned} T' \;&=\; \left(\mathcal{Q}^k \setminus \mathcal{V}_*\right) \cup \left\{w^i = z + (x_i^k - z_i)\mathbf{e}^i | z \in \mathcal{V}_*,\right. \\ &\qquad \left. i \in \{1, ..., N, ..., 2N\}\right\} \end{aligned} \tag{B.0.4}$$

After calculating the improper polyblock T', we can remove the elements $w^i = z + (x_i^k - z_i)\mathbf{e}^i$ which has an element $y \in \{z \in \mathcal{Q}^k | z \geq x^k\}$ such that $J(z, y) = \{i\}$, where $i \in \{1, \cdots, N, \cdots, 2N\}$. The new proper polyblock $\mathcal{Q}^{k+1}$ contains the remainder of the elements of T' after removing improper elements according to the rule above.

This procedure is repeated until either z^{k*} is feasible or $|cbv - cub| < \epsilon$, where cbv is the best objective function value found for the best feasible solution x^{k*} obtained so far, cub is the best objective function value found for the infeasible upper bound z^{k*} and ϵ is a small positive value which is user picked.

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