

Algorithm Design for Wireless Networks with Minimized Overhead Costs



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Statement of Originality

I hereby certify that the work embodied in this thesis is the result of original research done by me and has not been submitted for a higher degree to any other University or Institute.

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Abstract

Wireless network performance is typically affected by overheads arising due to unique challenges of the wireless medium. The broadcast nature of the medium implies that the channel is shared among all nodes, thereby prohibiting simultaneous transmissions from neighbouring nodes. This results in contention and collision overheads which get compounded with the increase in the number of nodes, posing great challenges to network scalability. In a multihop scenario, overheads are incurred due to challenges associated with traditional distributed systems. This is aggravated by contention overheads described earlier which make it difficult to disseminate information on a large scale.

In this work, the author studies the design of algorithms in wireless networks that incur zero or negligible overheads. The first problem addressed is differentiated channel access to provide quality of service (QoS) guarantees for real-time traffic. The author proposes a contention-tone based service differentiation scheme that leverages the benefits of out-of-band signaling to minimize collision probability. Using analytical and simulation results, the author shows that effective service differentiation can be achieved among the different traffic classes. The author addresses the issue of unfairness in two-way real-time traffic by using device specific differentiation in which the access point (AP) gets higher priority for channel access. The author shows that this results in equal opportunities for the AP and voice traffic, thereby resolving bottlenecks.

Converse to its effect on contention, the broadcast nature provides implicit benefits for data dissemination as all transmitted data is shared among neighboring nodes.

This feature has been referred to in existing literature as wireless broadcast advantage (WBA), which has been exploited for efficient protocol design to reduce transmission overheads. The author analyzes the performance benefits achievable purely by exploiting WBA in the context of network-wide broadcast. The analysis is focused on stateless broadcasting algorithms in which nodes decide on their forwarding behaviour based on information derived from overheard transmissions. The proposed approach categorizes the information available at the nodes into two distinct phases and identifies the performance benefits that can be attained from each. Based on insights obtained from the analysis, the author studies the design of stateless broadcasting algorithms in a multi-rate scenario. Subsequently, stateless algorithms that simultaneously reduce the broadcast latency as well as the percentage of forwarding nodes are proposed.

The author's third contribution concerns the self-organization of a wireless network so as to provide performance guarantees. A design that reorganizes the network in order to exhibit small world properties is proposed. The author studies the use of long range directional beams between nodes to achieve reduction in path length across the network. The proposed algorithm centers around a new centrality measure, which is defined as the wireless flow betweenness (WFB). WFB can be computed individually at nodes based on traffic flow information obtained from neighborhood transmissions. Subsequently, long range beams from a fraction of nodes results in significant reduction in the average path length of the network. Thus, the proposed design exploits WBA and minimizes explicit overheads. The proposed measure involves recursive computation at nodes that results in information being propagated over multiple hops, which is unlike the stateless broadcasting mechanisms mentioned previously in which the information is limited to the local neighbourhood.

Finally, the author studies the impact of WBA on the propagation of information over multiple hops. A setup in which nodes store and forward all information received implicitly from neighbourhood transmissions is considered and this results in the creation of a distributed cache across the network, which is termed broadcast cache. The

author then analyzes the growth of the broadcast cache in terms of the smallest set of transmissions in the network, which is defined as the set of non-altruistic transmissions. The author obtains the feasibility conditions which determine whether WBA can be effectively utilized depending on the flow characteristics.

The above problems are fundamental for a range of wireless network operations where it is imperative that overhead costs have to be minimized. The author's work also shows that the minimization of overhead costs can be achieved by a combination of algorithm design as well as the appropriate choice of parameters for the algorithm design.

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List of Abbreviations

Abbreviations	Expansions
AC	Access Category
ACK	Acknowledgement
AIFS	Arbitration Interframe Space
AN	Access Network
AP	Access Point
APL	Average Path Length
AR	Alerted Region
BOP	Broadcast Optimization Phase
BRP	Broadcast Recognition Phase
CBR	Constant Bit Rate
CDS	Connected Dominating Set
CT	Contention Tone
CTP	Contention Tone Protocol
CTS	Clear to Send
CW	Contention Window
DCF	Distributed Coordination Function
DIFS	Distributed Coordination Function Interframe Space
DTN	Delay Tolerant Network
DTOR	Directional Transmission Omnidirectional Reception
EAC	Expected Additional Coverage
EDCA	Enhanced Distributed Channel Access
ETS	Ensure to Send
ETT	Expected Transmission Time
ETX	Expected Transmission Count
FBC	Flow Betweenness Centrality
FR	Forwarding Region
HCCA	Hybrid Coordination Function Controlled Channel Access
IEEE	Institute of Electrical and Electronics Engineers

MAC	Medium Access Control
MCDS	Minimum Connected Dominating Set
MLB	Minimum Latency Broadcasting
MPEG	Moving Pictures Experts Group
MSG	Message
NS-2	Network Simulator 2
OBPS	Out-of-band Polling Scheme
OBS	Out-of-band Signaling
PCF	Point Coordination Function
PDR	Packet Delivery Ratio
QoS	Quality of Service
QUDG	Quasi-Unit Disk Graph
RAP	Rate Area Product
RBC	Routing Betweenness Centrality
RREQ	Route Request
RTS	Request to Send
SIFS	Short Interframe Space
SPBC	Shortest Path Betweenness Centrality
SRB	Savings Rebroadcast Ratio
TCP	Transmission Control Protocol
TR	Transmission Coverage Region
TSPEC	Traffic Specification
TTL	Time to Live
TXOP	Transmission Opportunity
UDG	Unit Disk Graph
ULA	Uniform Linear Array
VoIP	Voice over Internet Protocol
WBA	Wireless Broadcast Advantage
WCDS	Weighted Connected Dominating Set
WFB	Wireless Flow Betweenness
WLAN	Wireless Local Area Network

Chapter 1

Introduction

1.1 Research Objective

Overheads incurred in protocol design for wireless networks typically affect the overall network performance. These include both explicit protocol overheads as well as implicit ones arising due to the nature of the wireless medium. The author studies the design of algorithms that minimize the overheads incurred. As part of this dissertation, the author engages in both design and analysis of mechanisms for reducing overhead costs incurred in wireless network operations.

The research methodology adopted is as follows. First, the author identifies a set of problems which form the crucial aspects of wireless network operations and argue that suitable algorithm design for these problems to minimize overhead costs results in improved overall network performance. Subsequently, the author performs comprehensive literature survey for each of the problems outlined. This is followed by design and analysis of mechanisms that minimize the overhead costs and validation of the proposed research using extensive simulations.

1.2 Motivation and Scope

The proliferation of wireless networks has meant that they are now used for a wide range of applications. This necessitates a robust network architecture which can cater to varying application requirements. Overheads are typically the bane of such a network design as they affect all layers of the network stack.

Owing to the distributed nature of multihop wireless networks, implicit overheads result from network characteristics such as the size and density of the network. The information theoretic upper bound on the capacity of a multihop wireless network is obtained in the seminal paper by Gupta and Kumar [1]. Furthermore, increase in node density adversely impacts the channel access performance as shown in [2]. Additional overheads are imposed as a result of protocol specifications at the different layers. The theoretical performance limits for the IEEE 802.11 MAC are shown by the authors in [3]. The protocol overheads for reactive routing in ad hoc networks are studied by Zhou et al. in [4]. Similarly, in the context of geographic routing, the authors in [5] studied the overheads required for maintaining network information at the nodes.

As a result of overheads at the different layers, the final application throughput is generally much lower than the maximum capacity. To enable networks to scale to varying deployments and application requirements, it is therefore necessary to develop algorithms that minimize the overhead costs. The approach undertaken here is to identify a set of problems that are critical to the overall network performance. The author elaborates on the problems addressed and the corresponding overheads to be minimized.

1.2.1 Differentiated Channel Access

Handling traffic differentiation as part of channel access is necessary especially in crowded deployments so as to ensure performance guarantees for real-time and multimedia applications. The challenges involved in minimizing the impact of overheads is twofold. Firstly, such an algorithm needs to address the impact of overheads on chan-

nel access in general so as to maximize the total throughput. Subsequently, it needs to ensure that this throughput is differentiated suitably among traffic categories.

Existing literature has shown that the 802.11 MAC is heavily affected by overheads. The authors in [3] showed that a throughput upper limit and delay lower limit exist which bounds the performance even for extremely high physical layer bit rates. Their analysis considered the effects of fixed overheads which are part of the protocol specification. Moreover, collision overheads in the IEEE 802.11 *distributed coordination function* (DCF) also causes performance degradation in an environment where there is a high number of active stations [2]. An ideal design of a MAC layer protocol, therefore, needs to be able to scale to scenarios in which there are a high number of competing stations.

To address the service differentiation aspects of channel access, the IEEE 802.11e standard [6] defines an enhanced distributed channel access (EDCA), specifying a contention scheme to achieve differentiated channel access for streams of different types of traffic. While the performance analysis for the EDCA mechanism shows improved throughput [7] for higher priority data such as voice, the total throughput is well below the theoretical maximum and does not scale with increasing number of stations, primarily due to contention and collision overheads. In addition to differentiation between different traffic classes, a second aspect of service differentiation deals with addressing access point (AP) bottlenecks. This is particularly noticeable in two-way real-time traffic such as voice. Owing to the fact that the AP contends as a regular station but handles a much larger volume of traffic, the downlink rate is unable to match that of the uplink.

1.2.2 Network-wide Broadcast

Broadcasting is a fundamental operation required for a wide range of wireless network applications. Crucial tasks such as routing make use of network-wide broadcasting for route establishment. It is also essential for other applications such as information dissemination in sensor networks and for localization related tasks. Thus, efficient

design of a broadcasting strategy is critical to the overall network performance.

A straightforward and popular strategy for network wide broadcast is flooding, in which every node in the network acts as a forwarding node for the broadcast packet. Each node forwards the received packet to its immediate one hop neighbors. However, a multitude of issues with blind flooding in dense networks are identified as the broadcast storm problem by Ni et al in [8]. Increase in network size and density results in a higher number of transmissions due to flooding, which leads to redundancy and incur contention and collision overheads. Specifically in the context of reactive routing in ad hoc networks, [4] studied the impact of flooding as part of protocol specific overheads. This necessitates the need for design of broadcast algorithms that optimize the set of transmissions.

Algorithm design for optimizing broadcast performance has typically involved construction of a minimum connected dominating set (MCDS). However, such a construction requires knowledge of the network topology, which involves additional transmission costs. This impact on network resources is exacerbated with an increase in node density. Furthermore, it is difficult for such a design to adapt to dynamic topologies. An alternate class of broadcasting algorithms does away with the need for prior topological knowledge. Instead, nodes decide on their forwarding behavior based on information available from neighborhood transmissions.

The latter class of algorithms is clearly more attractive as they are stateless in nature [9] and exploit the wireless broadcast advantage (WBA). Wireless broadcast advantage (WBA) refers to the implicit sharing of information among neighborhood nodes as a result of the broadcast nature of the wireless medium. Algorithms that exploit WBA are thus likely to incur zero or negligible overheads. In order to develop efficient broadcasting algorithms, it is necessary to understand the performance benefits achievable as a direct result of the available information by exploiting WBA.

A second aspect of optimized broadcasting is to minimize the broadcast latency. The minimum latency broadcasting (MLB) problem has earlier been shown to be NP-Complete [10]. An attractive way of minimizing the broadcast latency is to leverage

the multi-rate capability of wireless networks. However, an implicit tradeoff exists between the two objectives of minimizing the latency and that of the number of transmissions due to the inverse relationship between transmission rates and their ranges [11]. As existing multi-rate broadcasting mechanisms are based on the connected dominating set (CDS) approach, the design of stateless algorithms needs to be investigated so as to minimize overheads while optimizing the tradeoffs between the design objectives mentioned above.

1.2.3 Self-Organization in Wireless Ad Hoc Networks

Given the diversity and scale of deployment of future wireless networks, an important requirement for ensuring scalability and reliability in networks is self-organization. An attractive model of reorganization to ensure performance guarantees in large scale networks is the small world network [12]. Small world networks are characterized by bounded path length between nodes along with high clustering coefficient achieved using a small number of shortcut links across the network. Small values of average path length imply that all nodes can reach each other in a few hops. The clustering coefficient measures the likelihood of nodes forming cliques in the network. A high value of the same implies that the network is well connected ensuring reliability. These characteristics are highly desirable for combining reliability with performance scalability in wireless ad hoc networks.

An important design paradigm for self-organization of wireless ad hoc networks is to make use of decentralized decision making at the nodes using localized information to achieve global properties [13, 14]. In order to realize small world behavior, the optimum set of nodes need to be identified for the creation of shortcuts. As the final objective is to minimize the path length over the entire network, the challenge here is to enable decentralized decision making at the nodes without global network information.

A popular measure of estimating node importance in the network is centrality [15]. However, a majority of definitions of centrality in the literature involve central-

ized computation and are not suitable for distributed implementation. Hence, it is necessary to identify a design in which nodes estimate their own importance in the network and make decisions regarding creation of shortcuts.

A specific challenge for creating small worlds in ad hoc networks is due to the length of the shortcuts. While traditional models of small world networks consider using random shortcuts between pairs of nodes, this is not possible in wireless scenarios as the edges between nodes are determined by their distances and the radio transmission ranges [16]. Existing mechanisms assume the presence of extra infrastructure such as nodes with higher capability radios or wired links. However, such assumptions limit the deployment to specific scenarios in which these conditions are met. In order to ensure performance guarantees for dynamic network deployments, a more generic solution is required.

1.2.4 Wireless Broadcast Advantage (WBA)

A popular way of reducing overhead costs is to exploit the wireless broadcast advantage (WBA). As mentioned earlier, WBA refers to the implicit sharing of information among nodes in a network as a result of the broadcast nature of the wireless medium.

Depending on the protocol design, exploiting WBA may either concern information regarding the immediate neighborhood or information propagated over multiple hops. The performance of network protocols that exploit the latter are determined by the extent of propagation of information as a function of WBA. Hence, for efficient protocol design, it is necessary to understand how information propagation can be facilitated by the broadcast nature alone.

The author aims to quantify WBA in a multihop scenario in terms of the information propagation over multiple hops.

1.3 Major Contributions

As part of the research work, the author addresses the problems identified in Section 1.2 as crucial to the overall wireless network performance. The approach adopted by the author includes both analysis and design of protocols which minimize overhead costs. The major contribution of this dissertation are given as follows:

1.3.1 Differentiated Channel Access

A multi-channel MAC protocol proposed in [17] which uses a slot length transmission burst called contention tone (CT) is used by stations to contend for channel access. Contention proceeds in a narrowband contention channel parallel to data transmission. A key aspect of the contention tone protocol (CTP) is that channel utilization is maximized resulting in throughput near the theoretical maximum. Furthermore, it is shown to scale in scenarios with high node density. These attributes make such a design ideal for service differentiation to cater to different traffic requirements.

The author proposes an approach which allows CTP to be flexible to cater to design requirements. The original design involved nodes contending with a constant CT transmission probability on the narrowband channel. This design is enhanced to allow nodes to compete with varying CT transmission probabilities. Additionally, a provision giving multiple chances to nodes is introduced, thereby allowing them to compete more than once. Therefore, the enhanced designed provides for dual degrees of freedom in terms of choice of parameters, which could be manipulated as per design requirements. Subsequently, in the context of the current problem, the author shows that effective service differentiation can be achieved by careful selection of these parameters. In summary, the major contributions for this section are to:

- Enhance the design of CTP to allow dual degrees of freedom.
- Identify optimal choice of parameters so as to achieve efficient bandwidth differentiation.

- Address the access point bottleneck problem by exploiting the degrees of freedom to differentiate between device roles.

The effectiveness of the proposed design is validated using both analysis and simulations.

1.3.2 Network-wide Broadcast

As outlined in Section 1.2.2, overhead costs in broadcasting can be minimized by exploiting information available at each node from neighborhood transmissions. Individual broadcasting mechanisms proposed in existing literature use various design guidelines that take advantage of specific aspects of this information. The author focuses on analyzing the performance benefits achievable purely as a result of this set of implicitly available information, i.e. without additional information which incurs transmission overheads.

In this regard, the author characterizes the implicitly available information as *network information* and develops a unified analytical model by characterizing the information available during the different stages of broadcasting. The model is then used to analyze broadcasting performance and obtain insights for algorithm design.

Subsequently, the author investigates the problem of multi-rate broadcasting. The relation between broadcasting at different data rates and how they impact the broadcasting performance are analyzed. The author obtains insights for the design of algorithms for multi-rate broadcasting. These insights are used to propose and compare stateless multi-rate broadcast algorithms.

The primary contributions in this section are summarized as follows:

- A unified performance analysis of stateless broadcasting obtained in terms of the *network information* at different stages of broadcasting. This model considers ideal channel conditions.
- Insights on broadcast algorithm design based on the above model in terms of feasibility conditions for network density and transmission costs.

- An extension of the above model to account for dynamic channel conditions and therefore provide further insights for robust algorithm design.
- Analysis and design of stateless algorithms for multi-rate broadcasting.

1.3.3 Self-Organization of Wireless Ad Hoc Networks as Small Worlds

The proposed approach for small world creation in ad hoc networks considers a generic setup in which any node in the network can be used for creation of shortcuts. The author leverages on the ability of directional beamforming to create long range shortcuts between nodes where the nodes do not differ with respect to their transmission capabilities.

The first part of the work does a simulation based analysis to identify the potential benefits and challenges of using randomized directional beamforming for creating shortcuts. Subsequently, the author uses these insights to design distributed algorithm for small world creation. In the process, the author defines a new measure of centrality for wireless ad hoc networks which can be computed in a completely distributed manner using only local information and incurring negligible overheads. This new measure of centrality is then used to propose a decentralized algorithm for small world creation.

The primary contributions in this section are as follows:

- A simulation based analysis of using randomized directional beamforming for small world creation in wireless ad hoc networks, with the achievable benefits and challenges identified.
- A new measure of betweenness centrality, defined specifically in the context of wireless ad hoc networks, that can be computed individually at the nodes to estimate their structural importance in the network. The proposed measure is distributed and lightweight in design.

- Distributed algorithm design for deterministic creation of directional beams for small world creation. The proposed algorithm makes use of the betweenness centrality defined earlier to enable decision making at the nodes.

1.3.4 Exploiting Wireless Broadcast Advantage as a Cache

This part of the work seeks to obtain a measure of the WBA in a multi-hop scenario. As mentioned earlier, the focus is to quantify the propagation of opportunistically received information over multiple hops. Considering a model in which all nodes store information received implicitly due to WBA, it is observed that this results in a distributed cache spread over the network. This is termed as *broadcast cache* and the growth of its contents is analyzed.

The main contributions in this section are summarized as follows:

- Analysis of the propagation of a symbol in terms of the growth of the contents of the *broadcast cache*.
- Establishing the conditions to determine whether exploiting WBA can result in performance improvements given the flow requirements.

1.4 Theme of Research

A common theme of the approach undertaken to solve each of the problems is to optimize performance by minimizing overheads involved in information sharing. A core focus is on WBA as it refers to the implicit sharing of information among nodes. However, since the nature of information depends on the problem itself, the manner in which WBA is exploited differs for each problem.

Traditionally, channel access mechanisms have been designed to take advantage of the broadcast property. In this case, the primary information required by nodes is the time slot in which collision free transmission can be achieved successfully. In the IEEE 802.11 DCF, nodes wait for random sequences of idle time slots, based on

neighborhood transmissions, before scheduling themselves. However, a major challenge here is to balance the waiting time required for estimation the channel state with that of actual packet transmission. As the waiting time increases with more collisions, performance degrades drastically with increase in node density. The degradation is particularly severe in the case of delay-sensitive applications such as real-time traffic. To optimize the resource allocation between sharing of channel state information and actual data delivery, the focus is on a multi-channel mechanism that separates the two operations. As outlined before, in the proposed service differentiation with CTP, a differentiated transmission schedule is determined based on channel access which proceeds in a separate narrow band channel in parallel with data transmission.

For the second problem of network-wide broadcast, the primary performance objective is to maximize the coverage while minimizing the number of transmissions. Typically, this requires topological information at nodes which can enable the creation of a connected dominating set (CDS). Instead, the discussion focuses on a class of algorithms in which information pertaining to broadcast effectiveness is derived from neighborhood transmissions. The information is local in nature, as it enables nodes to make locally optimal decisions on acting as forwarding nodes. Thus, this represents local exploitation of WBA even though the final application of network-wide broadcast is multi-hop in nature.

For the problem of self-organization of wireless networks, a core aspect of the approach taken is the definition of a betweenness centrality allowing nodes to identify their structural importance in the network. The proposed centrality measure is computed individually by nodes based on the set of overheard traffic flows. Instead of requiring nodes to store multi-hop network state which would incur overheads, the proposed approach involves recursive computation. The centrality value at each node is computed as a function of its neighborhood values, resulting in implicit propagation of information over multiple hops. Contrasted to the previous approach, this represents multi-hop exploitation of WBA as the information available at nodes gives an estimate of their structural importance in the entire network.

The final focus of the thesis is to study the implicit propagation of information over multiple hops by exploiting WBA. The motivation for such a study is that it represents the extent of propagation achievable in a network while incurring zero overheads. This in turn determines the performance benefits achievable by incurring zero overhead costs. Consequently, it also helps identify the tradeoffs involved in exploiting WBA.

1.5 Dissertation Organization

The rest of the dissertation is organized as follows:

Chapter 2 presents a comprehensive literature survey of each of the problems discussed in Section 1.2 and identifies the strengths and limitations of existing literature from the perspective of minimizing overhead costs.

Service differentiation using contention-tone based channel access is proposed and evaluated in Chapter 3. The proposed mechanism is shown to address bandwidth and device differentiation issues. A custom time-driven Python-based simulation framework is used to validate the results.

A comprehensive performance analysis of stateless algorithms is provided in Chapter 4. A unified analytical model is developed using insights drawn for the design of efficient stateless algorithms. Analysis and design of stateless multi-rate broadcasting algorithms are also looked into. Simulations are conducted in NS-2.

Chapter 5 explores the design of a distributed algorithm for self-organization of wireless networks as small worlds. A flexible design using directional antennas is proposed. Simulation results are obtained using MATLAB.

An analysis of wireless broadcast advantage (WBA) in a multi-hop network is presented in Chapter 6. The propagation of information over multiple hops is analyzed and conditions for efficiently exploiting WBA are identified. While the focus of the chapter is to provide a theoretical discussion on WBA, custom simulation results are also presented.

Chapter 7 concludes the dissertation and proposes future directions of research.

The list of publications by the author is presented in an unnumbered section succeeding chapter 7.

Chapter 2

Literature Review

In this chapter, the author surveys the existing literature for each of the problems outlined in Section 1.2 and studies the benefits and limitations of the current approaches and how they relate to the author's core motivation of alleviating overhead costs.

2.1 Differentiated Channel Access

As outlined in Section 1.2.1, there are two aspects to minimizing overheads for service differentiation in channel access. The first one deals with enhancing the performance of channel access itself so as to make it scalable while the second challenge is to meet the service requirements for different traffic categories.

Using multiple channels for channel access has been shown to increase the throughput per node [18]. As simultaneous transmissions can take place on separate, non-overlapping wireless channels, frame collisions are reduced significantly thus resulting in higher throughput. Since multiple channels are already available in the existing IEEE 802.11 systems, using them to increase the efficiency of the MAC protocol in the IEEE 802.11 WLANs has been indicated as a potential approach for high speed IEEE 802.11 WLANs. Given these benefits, multi-channel MAC protocols are ideally suited for providing service guarantees to QoS traffic.

2.1.1 Multi-channel MAC protocols

Multi-channel MAC protocols have earlier been classified based on their principles of operation by Mo *et al.* in [19]. *Single rendezvous* protocols in which the control information is exchanged on only one channel at a time are classified into Dedicated Control Channel, Common Hopping and Split Phase types. Another set of protocols allow multiple network interfaces to use several control channels in parallel to exchange control information and are known as the *multiple rendezvous* protocols. Slotted Seeded Channel Hopping (SSCH) [20] and McMAC [21] are two examples of such protocols.

The author concentrates on the class of dedicated control channel in the single rendezvous protocols. This class of protocols uses a dedicated channel for signaling while other channels are used for data transmission. The following paragraphs discuss some existing work in this class.

Nasipuri *et al.* suggest a multi-channel CSMA protocol with soft reservation in [22]. In the design, the total bandwidth is broken into N non-overlapping channels and one control channel. It is assumed that every node can listen to all channels. Any source that wants to transmit searches for a channel marked IDLE and transmits. Among the idle channels, the one used for the last successful transmission is used. This protocol is further extended in [23] to select the best channel based on signal power observed at the sender. A cooperative scheme based on the same model is suggested in [24] to reduce the interference to other receivers in the vicinity of the sender. However, these protocols require N transceivers per node, which make them expensive to implement. In [25], Kuang *et al.* suggest bi-directional data exchange with a view to improve the efficiency of TCP transfers.

Out-of-band signaling (OBS) with two separate channels using one for resource reservation and the other for data transmission has been introduced by Tantra *et al.* in [26]. In their work in [27] and [28], the authors show that the OBS scheme is effective in improving the performance of high-speed 802.11 WLANs. In the OBS scheme, two channels are used where a low bit rate signaling channel is dedicated for

managing channel access contention using an RTS/CTS like procedure as specified by DCF, and the actual data transmissions take place according to the point coordination function (PCF) on a high bit rate data channel. In the out-of-band polling scheme (OBPS) propose by Lim *et al.* in [29], a time division registration mechanism is used with OBS by dividing the signaling channel into slotted time. They also propose a priority scheme to enable service differentiation for different types of traffic.

Yang and Vaidya propose the use of pipelining techniques in the design of MAC protocols in [30]. By making use of a narrowband control channel for contention resolution, data transmission can proceed in parallel in a separate channel. In a partial pipelining scheme suggested in the same paper, the contention resolution takes place in two stages with winners from the first stage contending in the second stage. Here, the first stage of contention resolution takes place in the narrowband busy-tone channel while the second stage of contention resolution as well as data transmission take place in the data channel.

In the contention tone protocol (CTP) proposed in [17], a slot length transmission burst termed contention tone (CT) is used for channel assignment. A busy tone [31] can be used as the CT. Two channels are used in CTP. While data transmission takes place on the data channel, other nodes that have data to transmit access a narrowband slotted contention channel to compete for the next data transmission. It is shown that the contention resolution in CTP results in a channel utilization of 96%.

2.1.2 Traffic Differentiation

While multi-channel protocols provide opportunities for better utilization of the available bandwidth, recent advances in WLANs seek to provide service differentiation among different classes of data. The IEEE 802.11e standard [6] defines an enhanced distributed channel access (EDCA), specifying a contention scheme to achieve differentiated channel access for streams of different types of traffic. The standard defines four access categories (ACs), namely background (AC-BK), best effort (AC-BE),

video (AC-VI) and voice (AC-VO) in ascending order of priority. Effective service differentiation is achieved by configuring the EDCA parameters $AIFS[AC]$, $CWmin[AC]$, $CWmax[AC]$, and $TXOPlimit[AC]$ as per the access category. $CWmin[AC]$ and $CWmax[AC]$ denote the minimum and maximum contention window sizes for each AC respectively. The arbitration interframe space ($AIFS[AC]$) determines the time a node waits before transmission upon detection of an idle channel. Nodes with higher priority traffic are allowed to wait for shorter durations. Upon gaining access to the channel, nodes are allowed to transmit successive frames without contention up to a maximum duration determined by the transmission opportunity, denoted as $TXOPlimit[AC]$. While the performance analysis for the EDCA mechanism shows improved throughput [7] for higher priority data such as voice, the total throughput is well below the theoretical maximum and does not scale with increasing number of stations. This arises due to contention and collision overheads as well as a high fraction of idle slots which result in under-utilization of the channel.

A number of papers have previously concentrated on optimizing the DCF protocol parameters in order to provide effective service differentiation. The papers by Deng and Chang [32] and Romdhani et al [33] propose adjusting the interframe space and backoff generation function. In the two methods proposed by Bononi et al. in [34] and [35], they aim to reduce the number of contentions by estimating the optimal channel utilization. There are also proposals for multi-channel protocols with quality of service (QoS) support. One such proposal is the dual-channel request to send (RTS), clear to send (CTS), and ensure to send (ETS) suggested in [36]. Hamidian et al suggest enhancing EDCA to provide QoS guarantees in a single channel scenario [37] by reservation of TXOPs. The proportional delay differentiation mechanism suggested in [38] makes use of normalization of waiting times to achieve service differentiation among different classes of data.

Though service differentiation among different classes of data is an efficient way to meet the QoS requirements of real-time data, the effective throughput is still low in infrastructure networks. This is due to the unfairness among uplink and

downlink traffic in the presence of an access point (AP). As the AP handles all traffic in the network but contends as a regular station, downlink traffic experiences greater contention than uplink data from the stations. This imbalance has been studied earlier for different scenarios. The authors of [39] show the effect on TCP flows due to the unfairness between uplink and downlink streams. The authors of [40], [41] and [42] have reported similar behaviour in their respective studies. Solutions for achieving fairness include weighted fairness for the AP [43] and adaptively allowing the AP to transmit multiple data frames [44]. The disparity between uplink and downlink flows affects VoIP flows severely as the end-to-end performance suffers. Casetti and Chiasserini [45] propose service differentiation based on traffic direction in addition to the type while Gao et al. [46] suggest enhancing the EDCA mechanism by giving higher priority to the AP. Shin and Schulzrinne [47] use a Queue based priority for the AP. More recently [48], they analyze the various factors affecting VoIP capacity in WLANs. Hence, it is imperative that any service differentiation strategy designed to improve performance of real-time traffic also alleviates the bottleneck of data streams at the AP.

The literature discussed above concerns itself with service differentiation for distributed methods of channel access. An obvious solution to the access point bottleneck problem is to allow the AP to handle scheduling and prioritization of traffic, as with HCCA. Subsequent enhancements have examined optimal design of the HCCA parameters [49]. However, by design, centralized channel access mechanisms limit themselves to a scenario with a single access point which coordinates scheduling among the stations communicating directly with it. In crowded wireless scenarios where multiple APs are located in each other's vicinity, overlapping coverage areas can result in performance degradation. While APs can be tuned to different non-interfering channels, a limited number of such channels implies that the problem cannot be alleviated in crowded deployments. Though limited, existing literature has looked to address this scenario from the perspective of the point coordination function (PCF) [50] and the hybrid coordination function (HCF) controlled channel access (HCCA) [51].

2.2 Stateless Network-wide Broadcast

The simplest broadcasting mechanism is blind flooding. However, multiple issues with blind flooding in ad hoc networks have been identified by Ni et al. in [8]. As the network size and density increase, the number of nodes engaged in rebroadcasting a message increases as a result of flooding. This implies that transmissions from a majority of nodes end up being redundant since receiving nodes are likely to discard duplicate copies. Moreover, more nodes transmitting involves greater contention for channel access resulting in higher probability of collision, thereby degrading the overall performance.

A majority of research in broadcast algorithm design focuses on identifying a minimum connected dominating set (MCDS) [52] across the network. As determination of an optimal MCDS requires knowledge of the global network topology, distributed algorithms seek to optimize broadcasting of packets based on local neighbourhood information. However, dissemination of neighbourhood information involves additional communication overheads such as periodic broadcast of hello packets, affecting the overall network performance. The impact on network resources is exacerbated by an increase in node density. Furthermore, it is difficult for such a design to adapt to dynamic topologies. Therefore, it is necessary to consider alternatives that offer robustness of design while impacting minimal network resources.

An alternate class of broadcast algorithms relies on information derived from broadcast packets to determine node forwarding behaviour. They have been referred to as stateless broadcasting algorithms in existing literature [9] since they do not depend on any prerequisite neighbourhood information at the nodes. This makes them an ideal choice for purely distributed implementation. The primary design principle of stateless broadcasting algorithms is to maximize the additional coverage of each transmission within its neighbourhood. As it is difficult to accurately estimate the extent of coverage in a neighbourhood, nodes rely on information derived from overheard broadcasts to estimate the expected additional coverage (EAC). The wireless broadcast advantage (WBA) implies that transmission from any node is received by

all others within its transmission range. The information implicitly available from a transmission can therefore be utilized by all nodes receiving it. This set of implicitly available information is termed as the *network information*.

Stateless algorithms that use *network information* can broadly be classified as distance based and counter based schemes, the primary guidelines for which are derived from the observations in [8]. In distance based schemes, nodes estimate their EAC based on the distance to transmitting nodes. In the case of counter based schemes, the EAC is estimated based on the number of overheard transmissions. The authors in [8] analyze the problems of blind flooding, terming it as the *Broadcast Storm* problem. Subsequent efforts have looked to adapt the random delay values used in distance and counter based schemes. Chen et al. [53] differentiate the delays used by nodes based on their distance from the source. Heissenbuttel et al. [9] propose adapting the delay to network conditions. In addition to the observations in [8], important insights are also gleaned from the two references cited above. Firstly, the authors in [53] make use of the fact that the additional coverage is a function of both the distance to the overheard transmissions as well as the number of overhead transmissions. Secondly, since stateless algorithms are distributed, it is necessary for them to adapt to network conditions such as congestion arising from higher contention.

In spite of a rich literature of broadcasting algorithms, analytical treatment of the broadcast performance has been limited. Most of the research on stateless algorithms have focused on probabilistic flooding. Coverage and reliability analysis of blind flooding have been compared to that of probabilistic flooding by Viswanath and Obraczka in [54]. Mansour et al. also focus on the analysis of probabilistic flooding in [55], though they emphasize on a non-repetitive flooding scheme. More recently, random graph theory results are used to obtain optimal values of forwarding probability in [56]. Jun et al. obtain the delay performance of flooding using queueing models in [57]. Kuo and Liao analyze the effect of flooding on the hop count in [58]. Though their focus is not specifically on flooding, Foh et al. [59] obtain a distribution of hop lengths in a reactive routing protocol. The closest to the current focus in terms of

analyzing the effect of *network information* is [60]. However, the focus of the analysis in [60] is to compare individual schemes and not to understand the impact of available information in a generic sense.

The models used in the above mentioned literature analyze individual aspects of performance, and hence, cannot be used to gauge the impact of *network information* on broadcasting performance. For efficient algorithm design that maximizes the use of *network information*, it is necessary to develop a unified analytical understanding of stateless broadcasting.

Multi-rate broadcasting schemes in the existing literature have looked at construction of optimal broadcast trees in the network, based on the construction of connected dominating sets (CDS) for single-rate broadcasts. In [61], Chou et al. define the optimal broadcast tree in a multi-rate network as the minimum weight connected dominating set (WCDS), where the weight of an edge of the tree is the inverse of the transmission rate used. The authors show that such a tree can be constructed by maximizing the product of the transmission rate and the coverage area, denoted as the rate-area product (RAP). However, the construction of such a broadcast tree can be costly, as shown by the authors in [62]. A distributed algorithm is proposed by Guo et al. [63] which constructs a broadcast tree by identifying the optimum among multiple paths from the source to each node. None of the proposed algorithms takes advantage of *network information* to optimize multi-rate broadcasting performance by reducing the overhead costs.

2.3 Self-Organization of Wireless Ad Hoc Networks as Small Worlds

Quoting Dressler [14], self-organization in the context of ad hoc networks can be defined as: “Self-organization is a process in which structure and functionality (pattern) at the global level of a system emerge solely from numerous interactions among the lower-level components of a system without any external or centralized control.” In

the same paper, the inherent characteristics of wireless ad hoc networks are identified which makes it necessary to develop self-organized behavior. Important design paradigms for self-organization in wireless networks have been identified by Prehofer and Bettstetter in [13].

Small world is an attractive model for self-organization of wireless ad hoc networks as such a property ensures reliability and scalability. A popular model for creating a small world is by the addition of a small set of links, as shown by Watts and Strogatz [64]. Small world creation in wireless networks has first been investigated by Helmy in [16]. Simulation results are used to study the behaviour of wireless networks as a result of random addition of distance limited shortcuts. It is noted in [16] that, owing to the fact that wireless networks are spatial graphs rather than relational, shortcut links cannot be completely random as in [64]. Rather, the possibility of creating a shortcut link between two nodes is determined by the distance between them and the radio transmission range.

Subsequent research has looked into different strategies for creation of shortcut links. The authors in [65] and [66] consider a hybrid sensor network deployment consisting of a small set of wired links. The authors in [66] obtained theoretical results on the relation between the number of such wires required and the average path length. Guidoni et al. in [67] show that small world properties can be achieved in a heterogenous sensor network by using higher capacity nodes to create directed shortcuts towards the sink. In [68], Verma et al. propose three strategies for small world creation in wireless mesh networks in which nodes are equipped with two radios. Shortcuts are created between nodes using narrow beam directional links with the additional radios.

All the above mentioned strategies make use of additional infrastructure for creation of shortcuts. Such a design limits them to specific scenarios when the corresponding infrastructure, such as wires, additional radios and high capacity nodes, are present. A second limitation of the literature discussed above is that all of them consider at least some knowledge of the global network. Acquiring such information

incurs additional overheads, thereby impacting the network performance.

Thus, algorithm design for small world creation needs to ensure that it can be adapted for a wide variety of network deployments. Furthermore, as outlined in [13, 14], such a design needs to make use of purely locally available information.

2.4 Wireless Broadcast Advantage

Wireless Broadcast Advantage (WBA) is an inherent attribute of wireless networks which has been discussed extensively in the literature. Initially defined in the context of energy efficient broadcasting [69], WBA refers to the implicit sharing of information among nodes arising from the broadcast nature of the wireless medium. Transmission from a node can be received by all nodes within its range instead of just the destination.

Existing literature has looked to exploit this opportunistically received information to improve network performance. Mechanisms that exploit WBA can broadly be divided into single hop and multihop categories. Single hop mechanisms involve nodes making use of information concerning only their immediate one hop neighbours. Examples of such algorithms include cooperative MAC protocols [70, 71] in which source nodes optimize their transmission rates by making use of observed transmission rates between node pairs in its neighbourhood. Stateless algorithms for network-wide broadcasting [8, 60] make use of information available from overheard transmissions to optimize the set of transmitting nodes. Apart from optimizing local decision making at the nodes, WBA can also be exploited by propagating the implicitly received information over multiple hops. This has been shown to be particularly useful for combining flows over different paths to alleviate network bottlenecks, such as with network coding [72, 73]. Apart from packet flows themselves, WBA has also been exploited for propagating network specific information such as for localization [74]. As these examples show, WBA can be exploited to optimize network performance.

Existing research has sought to model wireless broadcast advantage in the context of specific applications. In the context of network coding, for instance, the broadcast

property has been modeled using hypergraphs for optimal scheduling [75] while others have explored the benefits of opportunistic listening for routing [76]. One work which is more closely related to the author's is [77] where the authors evaluate the scheduling performance as a result of the information available from the neighbourhood of a node, though this information is expected to be obtained through additional transmissions which incur protocol overheads. As far as the author knows, the proposed work is the first work that is purely devoted to obtaining a measure of the wireless broadcast advantage (WBA) in a multi-hop network. To the best of the author's knowledge, there is no existing work that obtains a suitable measure for the broadcast advantage itself.

Chapter 3

A Service/Device Differentiation Scheme for Contention-Tone Based Wireless LAN Protocol

In this chapter, the author addresses the two-pronged challenge of service differentiation through the design of QoS support for the contention tone protocol [17]. Contention-tone based channel access such as CTP seems particularly promising for such a design since it provides high throughput close to the theoretical maximum. The author introduces a new design based on CTP which enables differentiation between stations based on their traffic priority as well as roles, such as those of access point and wireless stations. With a view to achieving better effective throughput for real-time data such as voice, the author shows how this can be realized through careful selection of parameters. As the author's proposed approach is based on pure-CTP which guarantees greater than 95% successful contention resolution, an important feature is that it does not result in idle time slots. As a result of this, stations need not adapt their CT transmission probabilities depending on the channel state. As the optimal transmission probabilities can be calculated numerically for a wide range of network load, the system is more scalable. Compared to schemes which are based on exponential backoff, the actual data channel access probability of a station in CTP is

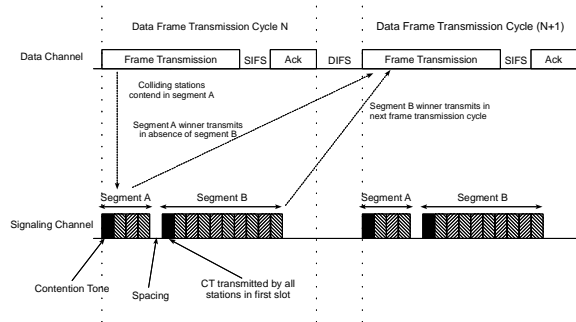


Figure 3.1: Illustration of contention-tone protocol.

equivalent to its probability of winning a contention round. This enables the service differentiation to be more deterministic. Analytical and simulation results show that this results in the required service differentiation being achieved while alleviating the access point bottleneck problem.

In the following section, CTP is revisited. Section 3.2 details the proposed design to include service and device differentiation in CTP. Performance analysis based on simulations is conducted in Section 3.3 to confirm the performance advantages. Section 3.4 is used to provide additional insights into the protocol performance as well as outlines a future direction of study following which important conclusions are drawn in Section 3.5.

3.1 Contention Tone Protocol (CTP)

This section gives a short description to the earlier proposed CTP which is necessary for the understanding of its QoS support described in the next section. Hereinafter, the earlier proposed CTP is termed as pure-CTP.

Pure-CTP proposes the use of a separate signaling channel with a separate circuitry for channel assignment purpose. The channel assignment is performed by contention. A contention tone (CT), which is a transmission burst lasting for a pre-defined slot period, is used for contention. The busy tone first proposed in [31] can be used for this purpose. However, instead of using it to indicate a busy channel, it is used here for resolving potential collisions involved in channel access. Use of

a separate channel dedicated to contention resolution allows pure-CTP to carry out channel assignment procedure concurrently with data frame transmission.

On the data channel, stations access the medium using DCF operation specified in the IEEE 802.11 standard [78]. When a station starts a data frame transmission on the data channel, it also starts a CT contention procedure on the signaling channel concurrently. The goal of the CT contention procedure is to distributively assign a winner among a group of contending stations. The CT contention procedure always ends before the complete DCF data frame transmission cycle so that a winner can be decided by the end of the data transmission cycle which includes a frame transmission followed by the acknowledgement. The winner from a contention procedure then transmits on the data channel in the next frame transmission cycle. An illustration of the pure-CTP operation during two consecutive data transmission cycles is shown in Fig. 3.1.

On the signaling channel, a DCF data transmission cycle is divided into two time segments, namely, *segment A* and *segment B*. The first segment, segment A is used to resolve collisions on the data channel. Segment B, on the other hand, is used by stations with backlogged data frames contending to gain access to the channel. Thus, segment A is only significant when the second segment fails to result in a single winner. In the event that two or more stations gain access to the channel in segment B, thereby causing a collision, they use segment A to decide on a winner among themselves. If both segments produce a winner each, priority is given to the winner in segment B.

The significance of segment A needs to be understood in the context of existing channel access schemes. In existing mechanisms such as DCF, owing to half-duplex communication, a packet collision can only be detected by a transmitting node after data transmission. Subsequently, each of the colliding stations contends again using new backoff counters for retransmission. In CTP, on the other hand, separation of contention and data transmission operations allows for early detection of collision using the segment A, with the provision that one of the colliding stations could

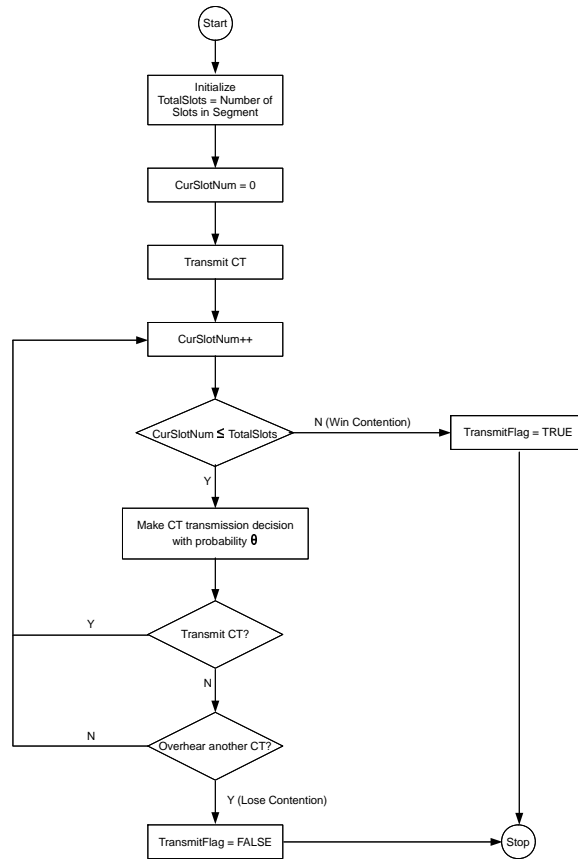


Figure 3.2: Flowchart illustrating the operation of a CTP segment.

be immediately rescheduled if no other station gets scheduled. Priority is given to winners of segment B as they have not been scheduled even once unlike the stations contending in segment A. Moreover, since any node transmitting on the data channel always transmits CT in segment A, giving priority to the same would make segment B redundant as the winning nodes in the latter would never get scheduled. Here, a case can be made for winners of segment A which have higher priority traffic than those of segment B. However, it needs to be noted that the information available to a node as part of the CTP operation does not include explicit information about other nodes it is contending with. Thus, differentiating between winners of segments A and B based on traffic priority would result in additional overheads required for sharing this information, thereby adversely affecting the performance of the entire network.

CT contention procedures for the two segments are similar. As shown in Fig. 3.1,

each time segment is subdivided into several mini-slots named contention slots. In the pure-CTP design, segment B consists of more contention slots than segment A in order to maintain a high success rate of generating a single winner.

In the CT contention procedure, a time segment always starts with a CT. In segment A, all stations currently transmitting on the data channel transmit a CT in the first slot. In segment B, stations not transmitting on the data channel but having data frames ready for transmission participate in the CT contention.

After the compulsory CT transmission in the first slot, for each following slot, each station participating in the segment may choose to either transmit a CT or listen to the signaling channel, but not both. When listening to the channel, if a station detects a CT, it loses the contention and stops transmitting CTs for the rest of the segment period. As the procedure progresses, more stations will lose the contention leaving fewer stations transmitting CTs towards the end of the segment period. The winner of the contention shall be the last station that transmits a CT. A single winner leads to a successful data frame transmission in the next transmission cycle. If there are two or more winners, a data frame transmission collision occurs, and the CT contention procedure for segment A will help resolve the collision. If both the segments produce a winner, priority is given to the winner of segment B. Hence, the winner of segment A only gets to transmit in the absence of segment B.

The CTP operation for any particular segment is illustrated using a flowchart in Fig. 3.2. In the figure, setting the transmission flag refers to whether the station proceeds to transmit data in the succeeding data transmission period. The number of slots in a segment would depend on whether it is segment A or B. As mentioned earlier, the number of slots would generally be higher for segment B as it would typically have more stations contending. Data transmission proceeds as mentioned earlier on the data channel.

It has been highlighted earlier that a unique feature of pure-CTP is the absence of idle slots. Unlike traditional contention schemes, nodes do not vary their contention probabilities on the basis of their estimation of channel state. Such a design arises

due to the fact that CTP separates the data transmission and contention operations. In exponential backoff based schemes, nodes attempt to reduce collision probability by waiting for random durations and subsequently increasing this duration upon encountering a collision. Hence, idle slots are essential for nodes to estimate channel state and subsequently schedule themselves. In CTP, on the other hand, the channel state information available at nodes pertains to that of the immediate scheduling status. This precise awareness of scheduling status implies that nodes need not decide on their transmission behavior based on an estimation of the channel state. Thus, idle slots are not created as part of the channel access mechanism, thereby resulting in high channel utilization. The only loss in channel utilization occurs due to collision, which is minimized by appropriate choice of CT transmission probabilities. In the existing literature, mechanisms based on deterministic backoff share a similar motivation to minimize the number of idle slots. However, such mechanisms are either implemented as part of centralized channel access such as HCCA [79] or are used in learning based schemes [80, 81] which use up a number of time slots to achieve collision free operation.

Nodes that are unsuccessful in a contention segment contend again during the next CT contention period, which takes place during the next transmission, with the same contention probabilities. Thus, as long as there is at least one node with a backlogged data frame, there are no idle time slots on the data channel. The only reduction in channel utilization occurs due to multiple winners from segment B, resulting in a collision. Since this is rare, very high levels of channel utilization have been achieved.

The narrowband contention tone channel should be completely orthogonal to the data channel so as to ensure zero or negligible interference. The channel bandwidth used for the contention tone channel could be similar to that of the busy tone described in DBTMA [82] and [31]. In DBTMA, for example, the busy tone is assumed to be implemented within a bandwidth of 10 KHz with a main data channel of 100 KHz. However, there are inherent tradeoffs involved in the choice of control channel bandwidth. Based on the discussion in [31], reserving 1% of the channel bandwidth can be an optimal design choice and a detection time of 5 μ s is sufficient in such a case.

Increasing the control channel bandwidth could marginally improve its performance as the signal can get detected earlier. However, this would have a negative impact on the overall channel capacity as reported in [31]. Hence, the author adopts a design choice of reserving 1% of the bandwidth for the control channel with the channel frequency spectrum being 22 MHz in IEEE 802.11 systems. A similar design has been shown to perform effectively by the authors of [83]. The transceiver architecture proposed in [84] and used in [82] can also be used here. Details of the pure-CTP operation and performance analysis are given in [17].

3.2 Contention Tone Protocol with Service Differentiation

3.2.1 Protocol Description

[17] shows that pure-CTP delivers promising performance. It uses the control channel in the MAC design to resolve contention and distributively schedule transmission. However, the design in [17] does not consider service differentiation. The need for service differentiation is obvious given the recent development of the IEEE 802.11e standard [6] that adds QoS support to the existing IEEE 802.11 MAC protocol with the specification of EDCA. The target of QoS support in EDCA is mainly the bandwidth differentiation among the four specified access categories (AC) [85].

The author makes use of two mechanisms to incorporate service differentiation as part of the CTP design. The author varies the CT transmission probabilities and makes use of multiple CT transmission chances to achieve prioritization among stations.

The author's design for CTP with service differentiation follows the design approach of EDCA. In particular, the proposed service differentiation adopts a distributed contention based approach. Instead of competing on the data transmission channel, the control channel is utilized for the contention and channel assignment.

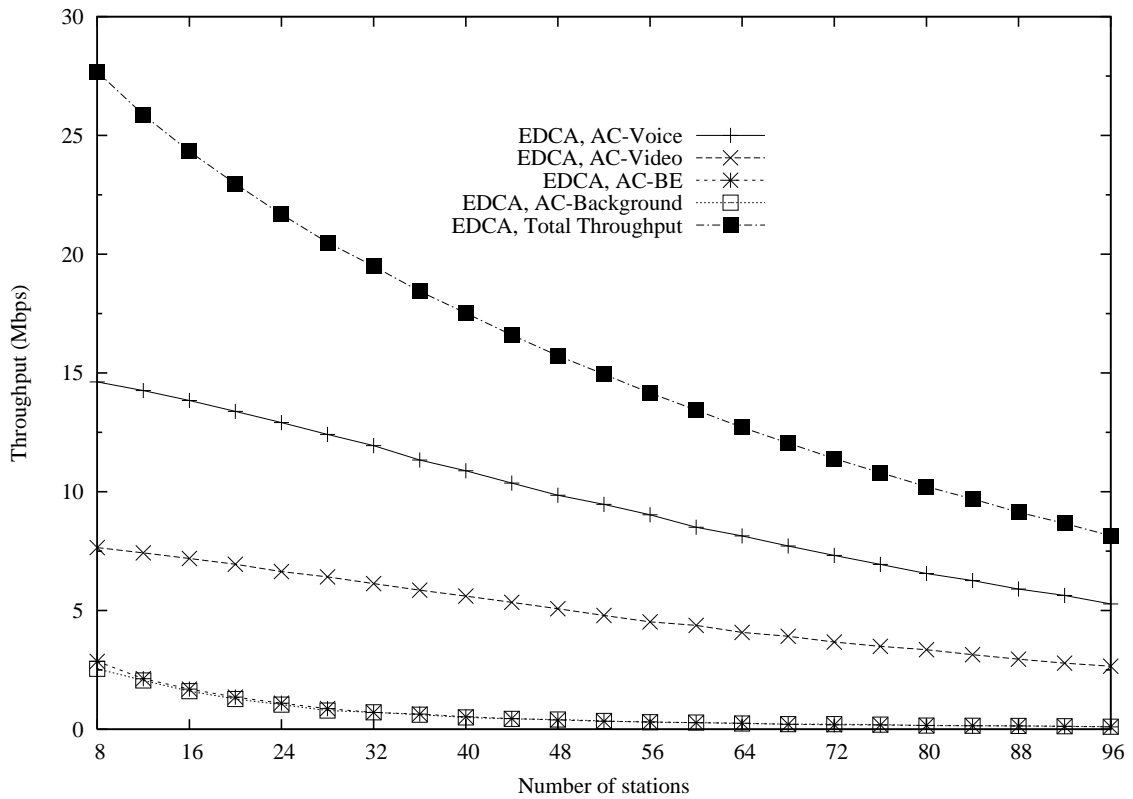


Figure 3.3: EDCA bandwidth utilization percentage.

The necessary extension for CTP is to include four ACs for the four defined traffic types.

The stations of CTP use CT to resolve contention on the control channel. In an earlier study conducted [17], a stations' probability of CT transmission in a particular slot affects its chances of winning that contention. A straightforward design to add four ACs is to have four different values of CT transmission probabilities, each for an AC.

Furthermore, the author incorporates additional ability to prioritize among stations with different roles such as access point and other nodes. The author adopts a design in which stations compete multiple times during a CT contention period by utilizing additional chances. This means a station continues to contend in a segment even after it has failed in particular slots, subject to a maximum number of chances. The immediate application is to alleviate access point bottleneck by balancing up-

link and downlink data streams. However, this can also be exploited as an additional degree of freedom for differentiation among stations, depending on the design requirements. Hence, for the ensuing discussion in the next subsection, consider a generic design in which every station makes use of additional chances during a contention segment. For a station belonging to AC i , the total number of chances is given as C_i . Thus, stations with higher priority traffic may be given higher number of chances, thereby improving its chances of winning a contention.

The motivation for proposing dual degrees of freedom for differentiation is to allow the protocol to be flexible to cater to design requirements. This is explained later in Sections 3.2.3 and 3.2.4. While varying the CT contention probabilities alone can satisfy the design requirements for bandwidth differentiation among different ACs, adjusting the number of chances can allow the author to suitably differentiate between roles. In the context of the current chapter, role differentiation between contending stations and the AP is addressed. Giving greater number of chances to the AP allows suitable balancing of uplink and downlink throughput of voice sessions.

Based on the earlier study in [85], a typical bandwidth differentiation of EDCA is given in Fig. 3.3. This performance result will be used as the guideline for the QoS support for CTP. In particular, the author follows the utilized bandwidth percentage of each AC for the QoS design, which is approximately 6:3:1.2:1 for the bandwidth utilization for AC3:AC2:AC1:AC0 under low traffic conditions. Using low traffic conditions as a baseline implies that the impact of contention and collision overheads on the bandwidth utilization percentages are minimized. Moreover, such a characterization seems to implicitly assume identical traffic characteristics for each of the ACs. This is because the objective of the proposed mechanism is to achieve differentiation in the success rates of channel access. The final throughput in a realistic scenario would depend on the traffic characteristics and is discussed in detail in 3.4.

While the design can easily provide service differentiation, it is also important to ensure that such a design maintains high performance which is the main objective of CTP. The assessment metric used to ensure high performance is to maximize the total

throughput of the system while maintaining appropriate bandwidth utilization ratios among the different access categories. In order to achieve this, the author aims to attain higher than 95% successful contention resolution as is promised by pure-CTP. This is formulated later as part of Eqn. (3.10) where the bandwidth differentiation objectives are also specified.

3.2.2 Analytical Model

Here, the author derives the expression for the probability of a CT contention segment to result in a single winner belonging to a particular access category. As with pure-CTP, the operation for both segments A and B are the same. Hence, the following discussion holds true for either segment. The total number of slots in a segment is denoted by ω , which depends on whether it is segment A or B. As discussed earlier, the winner from segment A only gets to transmit in the next data transmission cycle if segment B is absent.

Let N_i denote the total number of stations in the network belonging to AC i and let $K_{i,j}$ be the random variable of the number of stations belonging to AC i contending on the j -th slot. The quantity $K_{i,j}$ depends on the number of stations from all the ACs contending on the $(j - 1)$ -th slot. Of the stations contending in the $(j - 1)$ -th slot, all the stations which have transmitted a contention tone would be contending in the j -th slot. A station that does not transmit a contention tone in the previous slot would contend in the current slot if it has not yet exhausted its total number of chances or all other stations also do not transmit. Since the prerequisite for contention is that chances are not exhausted, it follows that the stations contending in the $(j-)$ -th slot have not exhausted their chances. The number of chances for stations belonging to AC i is denoted as C_i . Hence the probability density of $K_{i,j}$ can be expressed as

$$\begin{aligned}
 & Pr \left\{ \bigcap_{i \in \mathbb{L}} K_{i,j} = k_{i,j} \mid \bigcap_{i \in \mathbb{L}} K_{i,j-1} = k_{i,j-1} \right\} \\
 & = \begin{cases} \prod_{i \in \mathbb{L}} \theta_i^{k_{i,j}} + \prod_{i \in \mathbb{L}} (1 - \theta_i)^{k_{i,j}}, & \text{if } k_{i,j} = k_{i,j-1} \\ \prod_{i \in \mathbb{L}} \binom{k_{i,j-1}}{k_{i,j} - l_{i,j}} \theta_i^{k_{i,j} - l_{i,j}} (1 - \theta_i)^{k_{i,j-1} - (k_{i,j} - l_{i,j})}, & \text{otherwise} \end{cases} \quad (3.1)
 \end{aligned}$$

where θ_i is the CT transmission probability for AC i and $\mathbb{L} = \{0,1,2,3\}$ is the set of ACs. The term $l_{i,j}$ denotes the number of stations of AC i which do not transmit a CT but have not yet exhausted their number of chances, and hence, do not lose contention. Therefore, of the stations of AC i which are contending in the j -th slot, the number of stations which have transmitted a CT to contend are $(k_{i,j} - l_{i,j})$.

In order to obtain $l_{i,j}$, the author observes that the only stations which would lose contention upon not transmitting a CT in the $(j - 1)$ -th slot are the ones which would have exhausted exactly $C_i - 1$ chances till then. As mentioned earlier, in any preceding slot j' , a station exhausts a chance if it does not transmit a CT and at least one other station transmits. The probability of the occurrence of this is given as,

$$\nu_i = (1 - \theta_i) \left[1 - \prod_{i' \in \mathbb{L}} (1 - \theta_{i'})^{k_{i',j'}} \right] \quad (3.2)$$

Thus, the number of stations which have exhausted $(C_i - 1)$ chances would be a fraction of the number of stations contending in that slot,

$$n_{i,j-1} = \binom{j-1}{C_i-1} \nu_i^{(C_i-1)} k_{i,j-1} \quad (3.3)$$

since $k_{i,j-1}$ is the number of stations belonging to AC i contending in slot $(j-1)$. Since all other stations contending would not have exhausted their chances, the number of stations which would have exhausted less than $C_i - 1$ chances would be $k_{i,j-1} - n_{i,j-1}$. Thus, the number of stations of AC i which do not transmit a CT in the $(j - 1)$ -th

slot but still contend would be,

$$l_{i,j} = (1 - \theta_i)(k_{i,j-1} - n_{i,j-1}) \quad (3.4)$$

Note that, if $C_i = 1$, $l_{i,j} = 0$ at any time slot j as $n_{i,j-1} = k_{i,j-1}$. This corresponds to the original design of pure-CTP where a station loses contention whenever it overhears a CT from another station. Unconditioning Eqn. (3.1) results in

$$\begin{aligned} & Pr \left\{ \bigcap_{i \in \mathbb{L}} K_{i,j} = k_{i,j} \right\} \\ &= \left[\prod_{i \in \mathbb{L}} \theta_i^{k_{i,j}} + \prod_{i \in \mathbb{L}} (1 - \theta_i)^{k_{i,j}} \right] Pr \left\{ \bigcap_{i \in \mathbb{L}} K_{i,j-1} = k_{i,j} \right\} \\ &+ \sum_{i \in \mathbb{L}, m_i = k_{i,j} + 1}^{N_i} \left[Pr \left\{ \bigcap_{i \in \mathbb{L}} K_{i,j-1} = m_i \right\} \prod_{x \in \mathbb{L}} g_x(m_x) \right] \end{aligned} \quad (3.5)$$

where

$$g_i(m_i) = \binom{m_i}{k_{i,j} - l_{i,j}} \theta_i^{k_{i,j} - l_{i,j}} (1 - \theta_i)^{m_i - (k_{i,j} - l_{i,j})}.$$

and m_i is the number of nodes belonging to AC i contending in the $(j - 1)$ -th slot.

Success rate is defined as the probability of producing a single winner after a complete round of contention, composed of ω slots. The success rate for AC x having N_x stations is the probability that there is exactly one station from AC x contending on the last slot while there is none from any of the other ACs, i.e. $Pr\{(\bigcap_{i \in \mathbb{L} \setminus \{x\}} K_{i,\omega} = 0) \cap (K_{x,\omega} = 1)\}$. As per the proposed design, a station continues to contend in a segment until it exhausts all its chances. Thus, any station contending in the last slot ω would not have exhausted its chances till the previous slot. However, irrespective of the number of chances it has exhausted, the only scenario in which it does not transmit in the $(\omega - 1)$ -th slot but still wins the contention is that there are no other stations contending in that slot. Any other case would either lead to this station losing the contention or to a collision. Thus, the success rate, $S_3(N_0, N_1, N_2, N_3)$, for

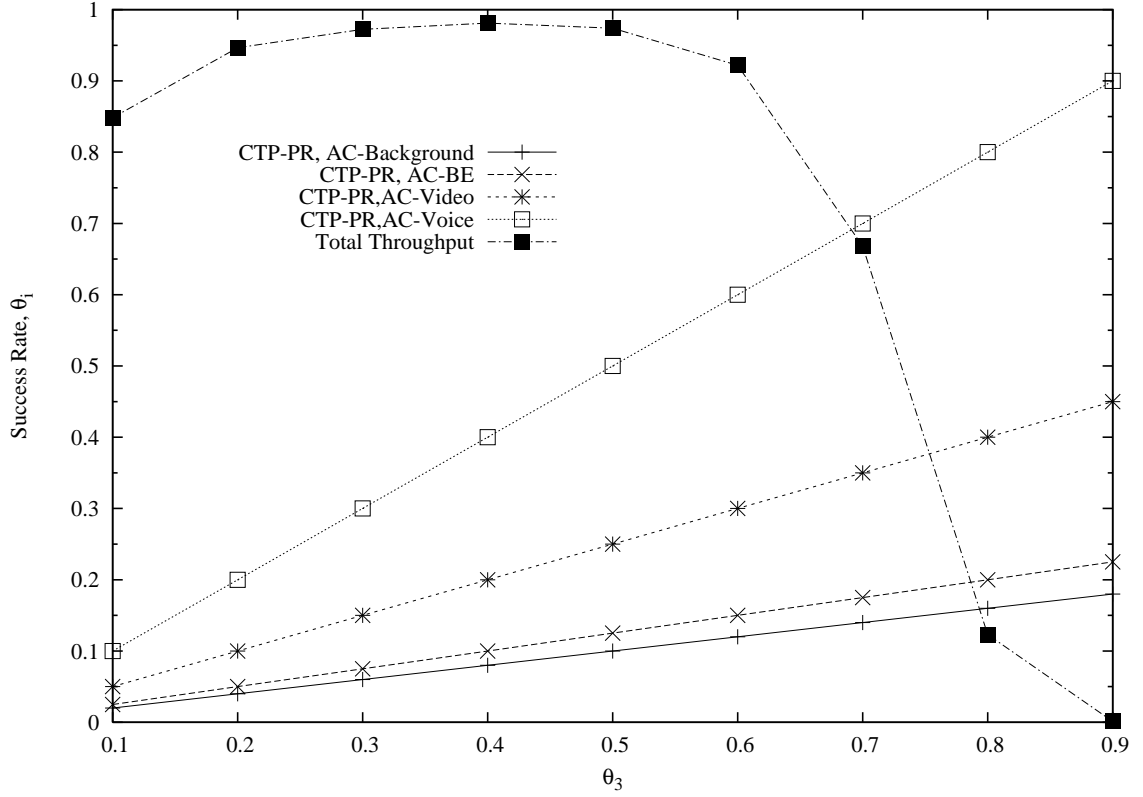


Figure 3.4: This figure plots results of θ_i vs θ_3 .

AC3 can be expressed as

$$\begin{aligned}
 & S_3(N_0, N_1, N_2, N_3) \\
 &= Pr\{K_{0,\omega-1} = 0, K_{1,\omega-1} = 0, K_{2,\omega-1} = 0, \\
 &\quad\quad\quad K_{3,\omega-1} = 1\} \\
 &+ \sum_{m_0=1}^{N_0} \sum_{m_1=1}^{N_1} \sum_{m_2=1}^{N_2} \sum_{m_3=2}^{N_3} [m_3\theta_3(1-\theta_3)^{m_3-1}(1-\theta_2)^{m_2} \\
 &\quad \times (1-\theta_1)^{m_1}(1-\theta_0)^{m_0} Pr\{\bigcap_{i \in \mathbb{L}} K_{i,\omega-1} = m_i\}]
 \end{aligned} \tag{3.6}$$

and the success rate for any other AC can be obtained in a similar manner. From the earlier discussion, CTP QoS design should support 6:3:1.2:1 for the bandwidth utilization for AC3:AC2:AC1:AC0. Since the success rate of an AC reflects the bandwidth utilization of that AC, it is then required that $S_3 : S_2 : S_1 : S_0$ also achieves 6:3:1.2:1 where $S_3 : S_2 : S_1 : S_0$ denotes the success rate ratio of ACs given a typical number of stations in each AC. The choice of parameters to achieve these design

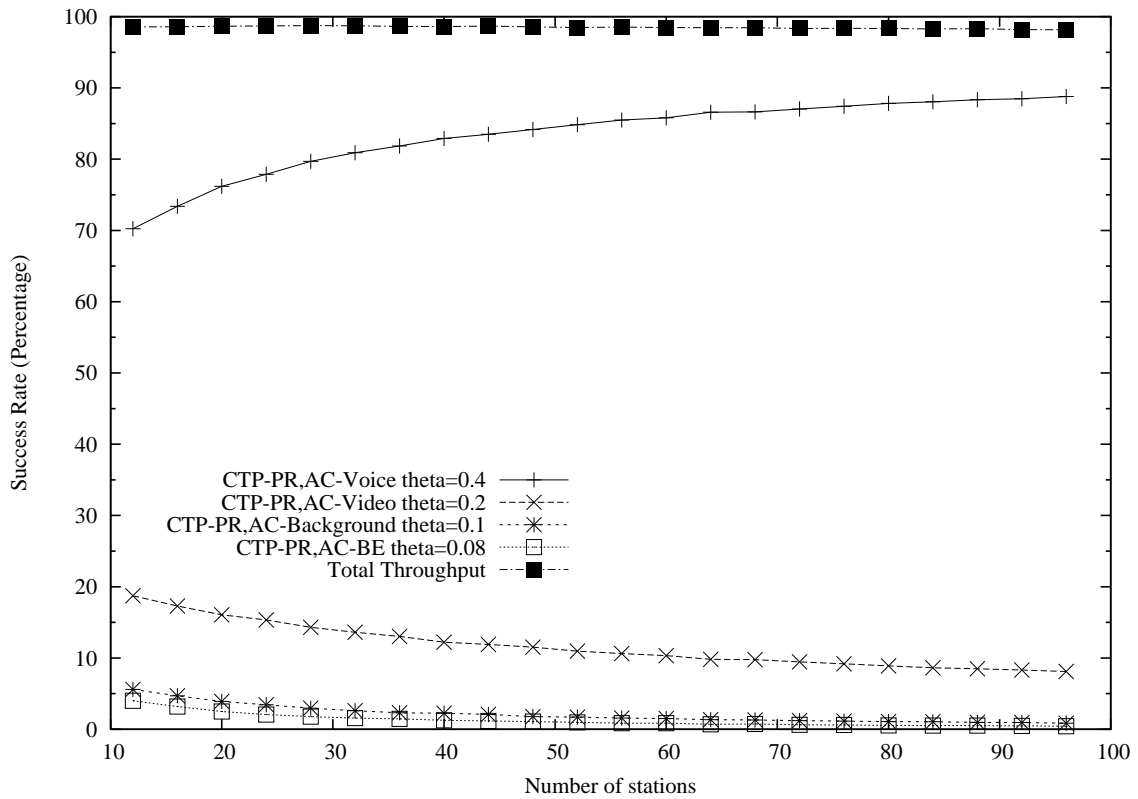


Figure 3.5: Success rate using constant CT transmission probability obtained analytically.

requirements is detailed in the next subsection.

The resulting success rate obtained above would determine the final throughput of the system. From [2], with P as the payload size, the maximum achievable throughput of DCF is

$$U_{DCF}^{max} = \frac{E[P]}{T_H + T_P + T_{SIFS} + T_{ACK} + T_{DIFS}}, \quad (3.7)$$

where T_H and T_{ACK} are the time durations required to transmit the header and acknowledgment respectively; T_P the time taken to transmit the payload; T_{SIFS} and T_{DIFS} denote the inter-frame spacing between transmitted frames [78].

Hence, for CTP, the throughput for each AC can be determined by

$$\begin{aligned} U_{CTP}(i) &= S_i(N-1) \cdot U_{DCF}^{max} \\ &= \frac{S_i(N-1)E[P]}{T_H + T_P + T_{SIFS} + T_{ACK} + T_{DIFS}}. \end{aligned} \quad (3.8)$$

where $S_i(N-1)$ is the success rate of AC i , with the term $(N-1)$ implying that there is a single winner and the number of stations contending in the second segment is $(N-1)$.

Applying Little's formula, the author further obtains the mean transmission delay performance of CTP with QoS. The transmission delay refers to the period between the time when a packet is generated and the time when the packet is successfully received by the destination. D_i is defined to be the mean transmission delay experienced by traffic of AC i , $i \in \mathbb{L}$,

$$D_i = \frac{\sum_{i \in \mathbb{L}} N_i}{U_{CTP}(i)}. \quad (3.9)$$

3.2.3 Parameter Settings for Bandwidth Differentiation

The success rate ratio between the ACs would depend on the values of θ_i and C_i chosen for each AC i . However, in the proposed design for service differentiation, the number of chances given to all ACs is being limited to 1. Thus, any time a station does not transmit a CT but overhears one, it loses the contention. Therefore, the service differentiation performance hinges on the range of CT transmission probabilities of different ACs, an appropriate choice of which would be sufficient to satisfy the design requirements.

The selection of the four CT transmission probabilities, $\theta_i, i \in \mathbb{L}$, may be based on the following optimization

$$\begin{cases} \text{Maximize} & S = \sum_{i \in \mathbb{L}} S_i \\ \text{subject to} & S_3 : S_2 : S_1 : S_0 = 6 : 3 : 1.2 : 1. \end{cases} \quad (3.10)$$

In Fig. 3.4, the author plots $\theta_i, i \in \mathbb{L}$ versus θ_3 that achieves $S_3 : S_2 : S_1 : S_0 = 6 :$

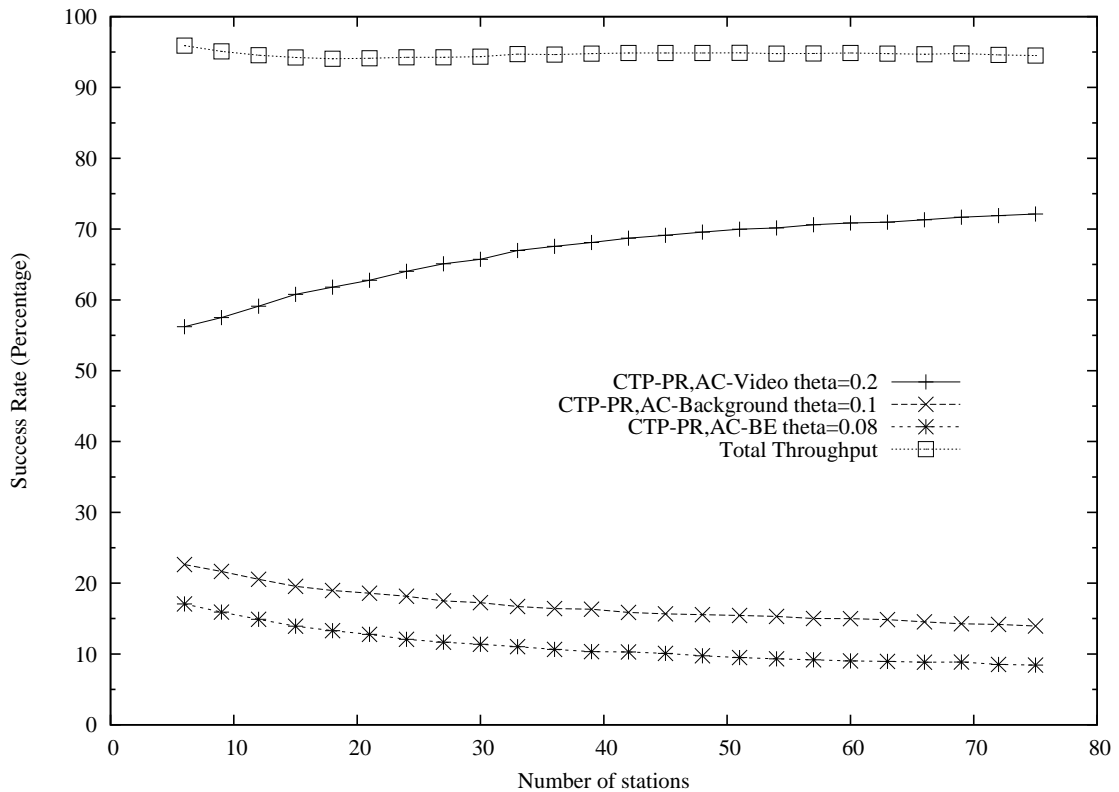


Figure 3.6: Analytical success rate of AC0-AC2 using constant CT transmission probability.

3 : 1.2 : 1 with $N_3 = N_2 = N_1 = N_0 = 25$. S is included in Fig. 3.4. As can be seen, choosing $\theta_i, i \in \mathbb{L}$ in the region where θ_3 is around 0.4 yields S that is near to its maximum value while maintaining the required bandwidth utilization percentages. The author also uses different combinations of N_i and consistent results have been obtained. Hence, only a combination is shown here for illustration purpose.

Based the above results, the author chooses 0.4, 0.2, 0.1 and 0.08 for the CT transmission probabilities for AC3, AC2, AC1 and AC0 respectively that satisfy the requirements given in Eqn. (3.10). The numerical results for the bandwidth utilization of each AC for the chosen setting is plotted in Fig. 3.5. The results show the achievement of targeted bandwidth utilization percentage for each AC. The overall success rate is maintained above 95%.

Further tests are conducted to study the efficiency of the proposed design. In one test, the network contains no AC3 traffic, which corresponds to the situation

where there is no user running voice application. Based on the author's design and varying the number of stations for other ACs, the author plots the results in Fig. 3.6. Interestingly, the overall success rate drops appreciably. The main reason is that traffic belonging to access categories AC2 and below use low values of CT transmission probability. This results in the inability to achieve a single winner as the probability that no station transmits in any particular slot is higher (refer to the second term in the first part of Eqn. (3.1)). Subsequently, stations continue to contend in the rest of the slots in the segment. As a result, the probability that the contention period results in more than one winner, and therefore a collision, is higher. Thus, the lack of aggressiveness in CT transmission can result in underutilization of the contention operation and therefore lower overall success rate. To address this problem, an alternative design for the CTP QoS support is required.

Past experiences of similar operations in other domains [86] indicate that constant values for CT transmission probabilities do not always produce an optimal result. The optimal result here refers to the high success rate. While high success rate has been achieved in the case of pure-CTP, it can be seen that this design fails to produce satisfactory result when multiple traffic types are considered. A possible alternative for the proposed design is the introduction of a function that describes the probability values over contention slots. In the author's proposed design, the author seeks the simplest possible function that achieves maximum success rate.

The function chosen is a simple linear function that can be specified by the starting and ending probabilities for the first and the last contention slots respectively. Each AC defines such a linear function, and all four functions share the same value for their ending probability. For the starting probability of the function, intuitively, a higher prioritized AC requires a higher starting probability. To determine the exact values for the starting probabilities for all four functions under the constraint that the projected percentage of bandwidth utilization of each AC must match that of the EDCA, the author first proceeds with modeling the proposed method.

In the case of such a linear variation, the value of the transmission probability θ is

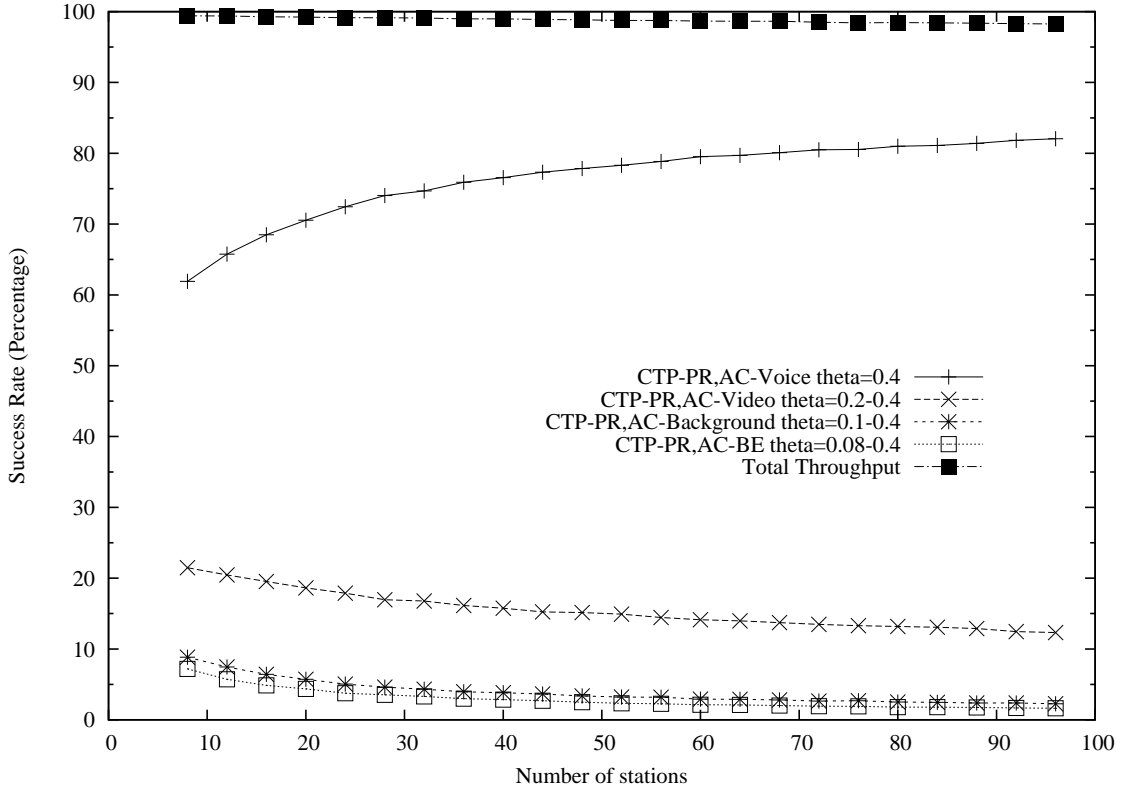


Figure 3.7: Analytical success rate using linear CT transmission probability.

varied across the contention slots. For an AC, say AC i , the transmission probability in a slot j is given by $\theta_{i,j}$. The quantity $\theta_{i,j}$ has an initial value of θ_i^{min} , and is increased after every contention slot by a constant value $\delta_i = (\theta_i^{max} - \theta_i^{min})/\omega$ where θ_i^{max} is the ending probability for AC i and ω is the number of slots in a contention round.

Thus, the CT transmission probability for AC i in the j -th slot is given by

$$\theta_{i,j} = \theta_i^{min} + \delta_i(j - 1). \quad (3.11)$$

Replacing θ_i with $\theta_{i,j}$ in Eqns. (3.1)-(3.6) and repeating the calculation yields the success rates for all ACs. The search for solutions for CT transmission probabilities

for all ACs that satisfy the following is repeated:

$$\begin{cases} S_1 : S_2 : S_3 : S_4 = 6 : 3 : 1.2 : 1 \\ S_1 + S_2 + S_3 + S_4 \geq 95\% \end{cases} \quad (3.12)$$

Using the linear variation for the CT transmission probabilities, the author finds that the setting where

$$\begin{cases} \theta_3^{min} = 0.4 \\ \theta_2^{min} = 0.2 \\ \theta_1^{min} = 0.1 \\ \theta_0^{min} = 0.08 \\ \theta_i^{max} = \theta_3^{min}, i \in \mathbb{L} \end{cases}$$

consistently yields more than 95% success rate even at higher loads. In Fig. 3.7, the success rates for all ACs using the chosen linear variation for the CT transmission probability are plotted. As can be seen from the figure, not only can a high success rate be achieved, but also the projected bandwidth utilization percentages for all ACs can be maintained. As shown above, the parameter values have been chosen to suit the design constraints. They can similarly be tuned to fit any other design.

3.2.4 Parameter Settings for Balancing Uplink vs Downlink Performance

Many recent research works have indicated that the current bandwidth differentiation QoS mechanism in WLANs leads to communication bottleneck at the AP when applications produce two-way real-time symmetrical traffic. These applications often connect a local user at a WLAN to a remote client from the Internet via the AP of the WLAN. The AP serving as the gateway between the local users and remote users must handle all downlink traffic in the WLAN.

However, in the protocol design, including that of the IEEE 802.11e MAC protocol, an AP is viewed as an ordinary station in the MAC design. Since the MAC protocol does not differentiate between ordinary stations and the AP, the AP shares the same

bandwidth with all other stations. For example, given N simultaneous sessions each utilizing the same bandwidth, the AP needs to handle N times more bandwidth than other stations. The failure to grab a bigger share of the bandwidth results in it being a bottleneck. The presence of this bottleneck problem in the IEEE 802.11 WLANs is reported in [39] - [48] and in [87]-[88], some of which have been discussed earlier.

Since the proposed protocol operation inherits the DCF which is part of the IEEE 802.11 MAC operation, CTP also suffers from the above AP bottleneck problem. One common and effective approach to solve the AP bottleneck problem is to provide appropriate device differentiation in the design of the QoS support, which the author has adopted. The author uses the design choice of additional chances for the AP so as to enable it to utilize a higher bandwidth for VoIP sessions in particular.

The AP periodically monitors AC3 traffic to detect voice session. When the AP receives AC3 traffic from a station, it considers that this station has started a voice session. The AP then adds a record containing the MAC address of the station to its voice session list. Each record has a predefined time to live (TTL) to specify its lifetime. If the AP detects a voice session that is already recorded in the list, it simply resets the TTL of the record. This operation allows the AP to maintain the list of active voice sessions.

The AP needs to ensure that it can serve all detected voice sessions. Given N_3 voice sessions, the AP must be able to utilize N_3 more bandwidth than that of each station running a voice session. As part of the proposed design for CTP with service differentiation, the author has limited the number of CT transmission chances in a segment (CTC) for all categories to 1, thereby negating the effect of this feature. Here, the author proposes allowing the AP a higher value of CTC, while for all other stations it still stays at 1. Precisely, the AP uses the same CT transmission probability as that of AC 3 but uses a higher number of chances. Thus, during the CT contention, if the AP detects a CT when it itself decides not to contend, it increments the number of chances it has exhausted. It loses contention upon exhausting the maximum number of chances. The bandwidth utilization of the AP can be controlled by adjusting the

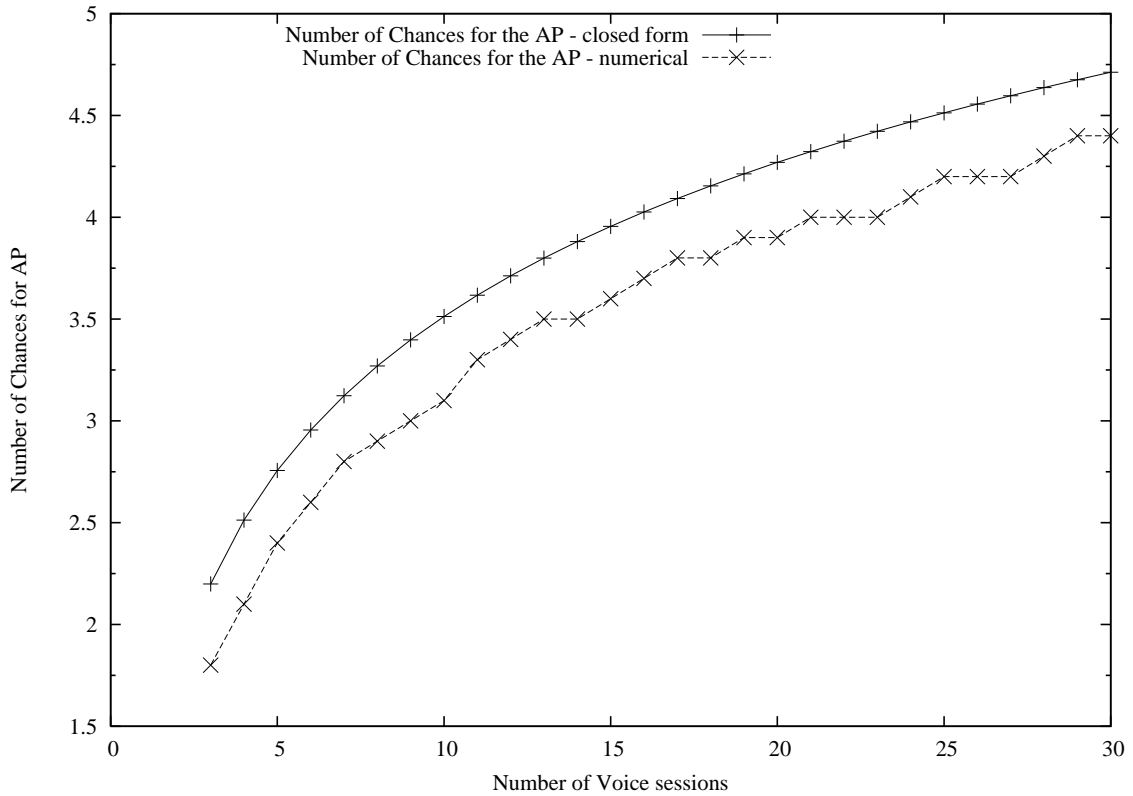


Figure 3.8: Number of chances vs Number of voice sessions (both numerical and closed form).

value of CTC assigned to it. The number of chances given to the AP depends on the number of voice sessions as the AP utilizes N_3 more bandwidth than each station running a voice session given N_3 voice sessions.

To fulfill the proposed design, it is necessary to develop the relationship between the number of voice sessions and the corresponding number of chances for the AP. Therefore, the success rate of the AP when it contends in a CT segment needs to be identified. When the AP detects a transmission while listening to the channel in any particular slot, it gets additional chances to contend subject to a maximum of number of chances, C_{AP} , while other stations lose contention in such a scenario since they only have a single chance. Thus, the probability that the AP contends in any particular slot j depends on the number of chances it has exhausted until that slot, given as $\gamma_{AP,j}$. The AP can contend as long as $\gamma_{AP,j} < C_{AP}$. Following the same analysis and reasoning as in Section 3.2.2, the success rate of the AP, or the

probability that a contention round ends with the AP as the winner can be obtained as follows:

$$\begin{aligned}
 & \hat{S}_{AP} \\
 &= Pr \left\{ \bigcap_{i \in \mathbb{L}} K_{i,\omega-1} = 0, \gamma_{AP,\omega-1} < C_{AP} \right\} \\
 &+ \hat{\theta}_{AP} \sum_{i \in \mathbb{L}, m_i=1}^{N_i} \left[(1 - \theta_0)^{m_0} (1 - \theta_1)^{m_1} (1 - \theta_2)^{m_2} \right. \\
 &\left. \times (1 - \theta_3)^{m_3} Pr \left\{ \bigcap_{i \in \mathbb{L}} K_{i,\omega-1} = m_i, \gamma_{AP,\omega-1} < C_{AP} \right\} \right]
 \end{aligned} \tag{3.13}$$

Thus, the number of chances utilized by the AP, C_{AP} would depend on the success rate of the AP as well as that of voice traffic. Thus, the choice of C_{AP} would have to be conditioned on,

$$\hat{S}_{AP} = S_3(N_0, N_1, N_2, N_3) \tag{3.14}$$

Based on this modeling, the author should be able to achieve device differentiation by adjusting the values of either θ_{AP} or C_{AP} . Intuitively, in the first case, θ_{AP} should be much higher than the rest of the stations for the targeted device differentiation to be realized. However, from the earlier discussion in section 3.2.3, it can be inferred that this would adversely affect the total throughput of the system, as illustrated in Fig. 3.4. Hence the author converges on a design which chooses a suitable value of C_{AP} while keeping $\theta_{AP} = \theta_3$.

From Eqns. (3.13) and (3.14), the author can compute numerically the number of chances to assign for the AP given the number of detected voice sessions. However, relying on numerical results confines the operation to where the numerical results are well prepared. To overcome this shortcoming, the author further develops the closed form results describing the relationship between C_{AP} and the number of detected voice sessions.

Consider N_3 voice sessions. Since each voice session consists of an uplink and a downlink transmission, there are $2N_3$ transmissions where each voice station contributes one and the AP contributes N_3 transmissions. In other words, each voice

station utilizes $\frac{1}{2N_3}$ and the AP utilizes $\frac{1}{2}$ of the total bandwidth consumed by voice sessions.

Let η be the number of CT slots required to generate a single winner out from N_3 stations and the AP. It is noted that if N_3 is large, the probability that none of the stations transmits in a contention slot is low (less than 0.5% if there are more than 10 stations), and this rare event can be ignored. Thus, a station which will become the ultimate single winner of a particular CT contention must transmit η consecutive CTs. Recall that AC3 uses a constant CT transmission probability, which is θ_3 , then

$$\theta_3^\eta = \frac{1}{2N_3}. \quad (3.15)$$

For the AP, let c be the number of assigned chances (C_{AP}), and it will utilize $\frac{1}{2}$ of the total AC3 bandwidth. Since the AP has $c - 1$ chances of not transmitting a CT but still becoming the winner of a particular CT contention, then

$$\theta_3^{\eta-(c-1)} = \frac{1}{2}. \quad (3.16)$$

Solving for Eqns. (3.15) and (3.16) yields

$$c = \frac{\ln(1/N_3)}{\ln \theta_3} + 1 \quad (3.17)$$

In Fig. 3.8, the author plots the numerical and closed form results for the number of chances versus the number of voice sessions. It is noted that the proposed design allows for non-integer numbers of chances. In the implementation, to achieve the actual c chances, it is done by assigning an appropriate probability, say q , where with probability q (resp. $1 - q$), the AP uses $\lfloor c \rfloor$ (resp. $\lceil c \rceil$) as the number of chances in a CT contention period to achieve the targeted c chances. In other words,

$$c = q\lfloor c \rfloor + (1 - q)\lceil c \rceil.$$

The probability, q , can be easily obtained by solving the above formula.

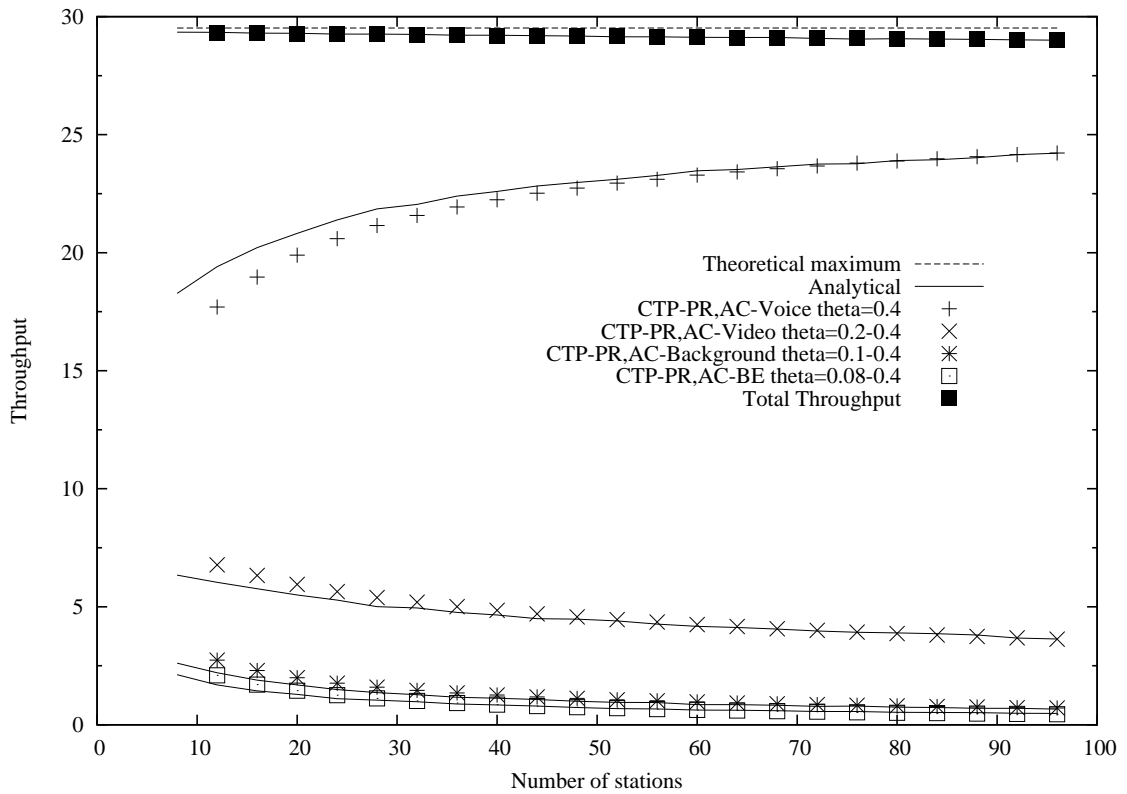


Figure 3.9: Simulation results for saturation throughput of all ACs.

In Fig. 3.8, the author plots the number of chances given to the AP as obtained numerically as well those obtained from the closed form described above. The number of chances obtained using the closed form seems significantly more than that obtained numerically. In order to verify the effectiveness of the closed form expression, the author performs extensive simulation based on the closed form results instead of the numerical results to compute the number of chances given to the AP. As would be seen later, the bandwidth utilization of the AP is comparable to that of voice traffic, which is the targeted performance. All the simulation results are discussed in detail in the next section.

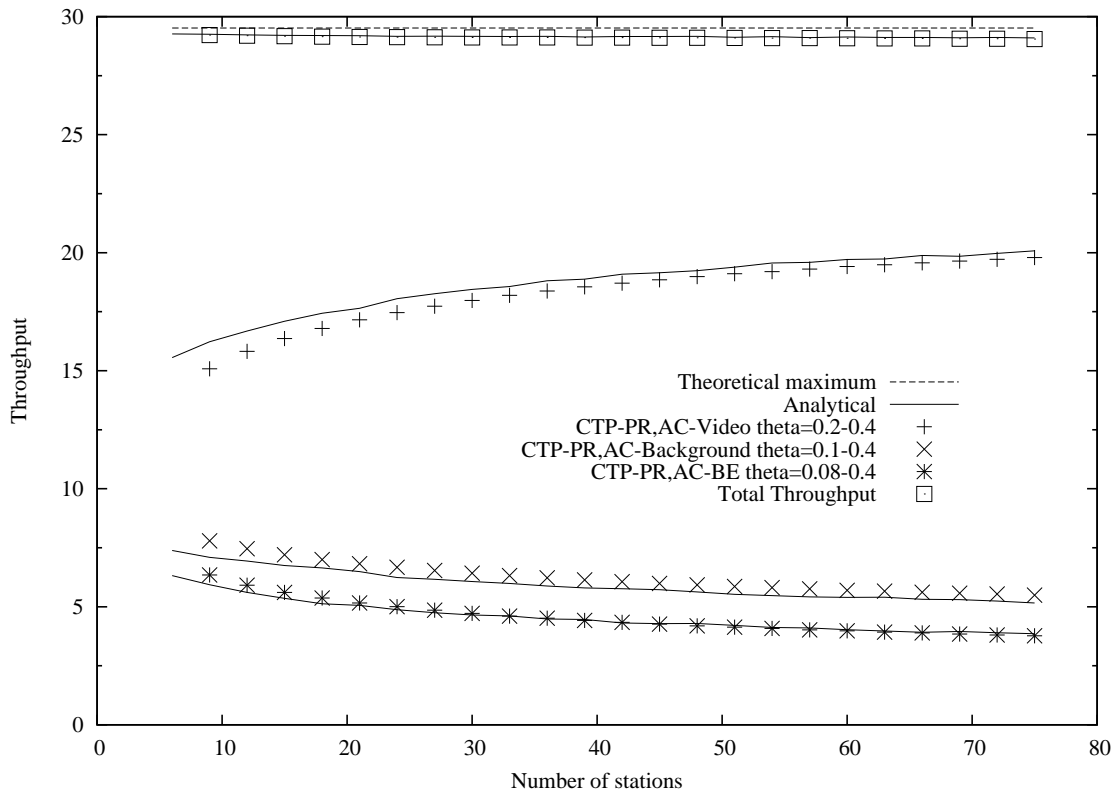


Figure 3.10: Simulation results for saturation throughput of AC0-AC2.

3.3 Simulation Results

The author presents simulation results to validate the performance of the proposed contention tone protocol with service differentiation. The simulations are carried out using a custom event-driven simulator. The simulator has been used to correctly reproduce the results of pure-CTP. The accuracy of the simulator has earlier been verified by using it to validate IEEE 802.11 saturation throughput as part of the works in [89, 90]. The authors of [91] compared their EDCA saturation analysis, verified using OPNET, to that in [85], which has been verified using the same simulator as the one used here, thus providing an implicit validation of the simulator's accuracy. All the simulations have been carried out using a packet size of 1000 bytes and a data rate of 54 Mbps under saturation conditions. Each simulation for a particular number of users is run for 10 iterations with a 95% confidence interval and the mean values are shown in the results.

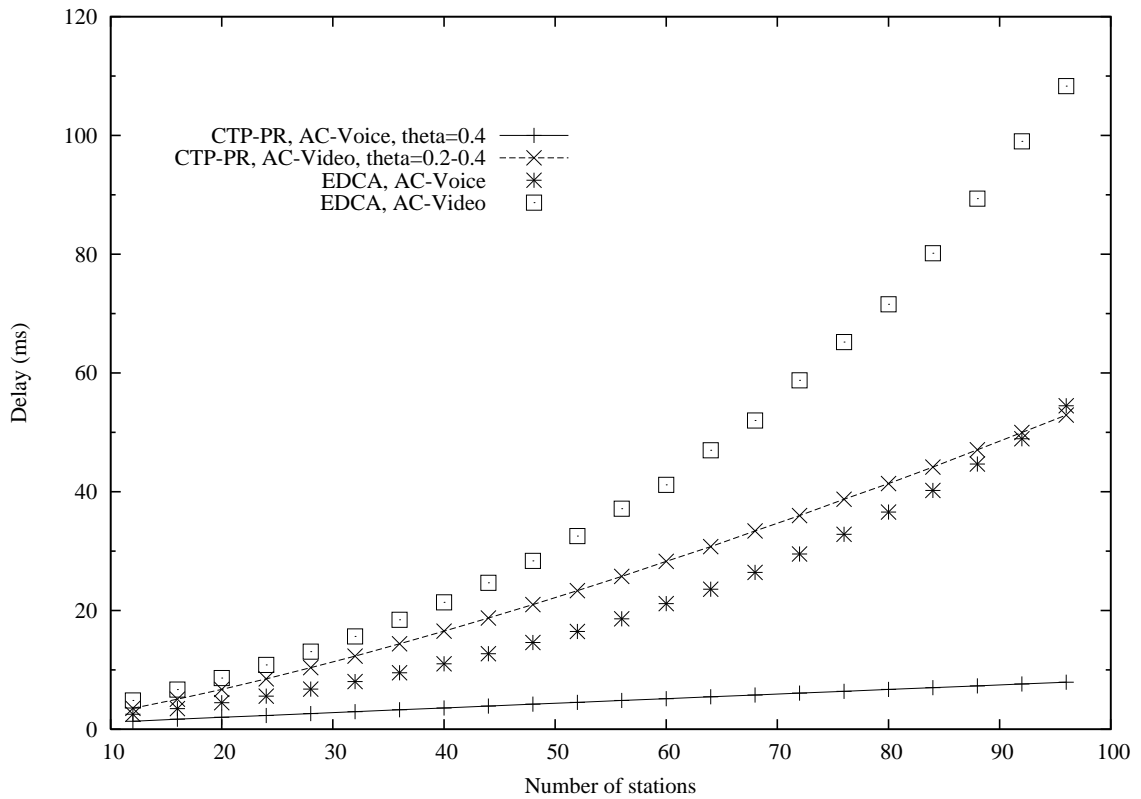


Figure 3.11: Simulation results for the mean delay performance.

Fig. 3.9 compares the saturation throughput performance of CTP with QoS, which is referred to as CTP-PR, to the theoretical maximum. The author chooses saturation traffic in order to closely map the success rates of gaining channel access to the final data transmission performance. The projections of the corresponding analytical measurements are also presented. Comparing with the IEEE 802.11e performance depicted in Fig. 3.3, it can be seen that while EDCA suffers from throughput drops as the number of stations increases, CTP-PR maintains high throughput even with a high number of stations. This is as expected because of the high success rate at all densities, which is itself a result of low probability of collisions.

To verify that the proposed design remains effective when the WLAN does not carry all classes of traffic, the author tests the scenario where AC3 traffic is absent. This result is shown in Fig. 3.10. The overall throughput in this scenario remains close to the maximum achievable throughput.

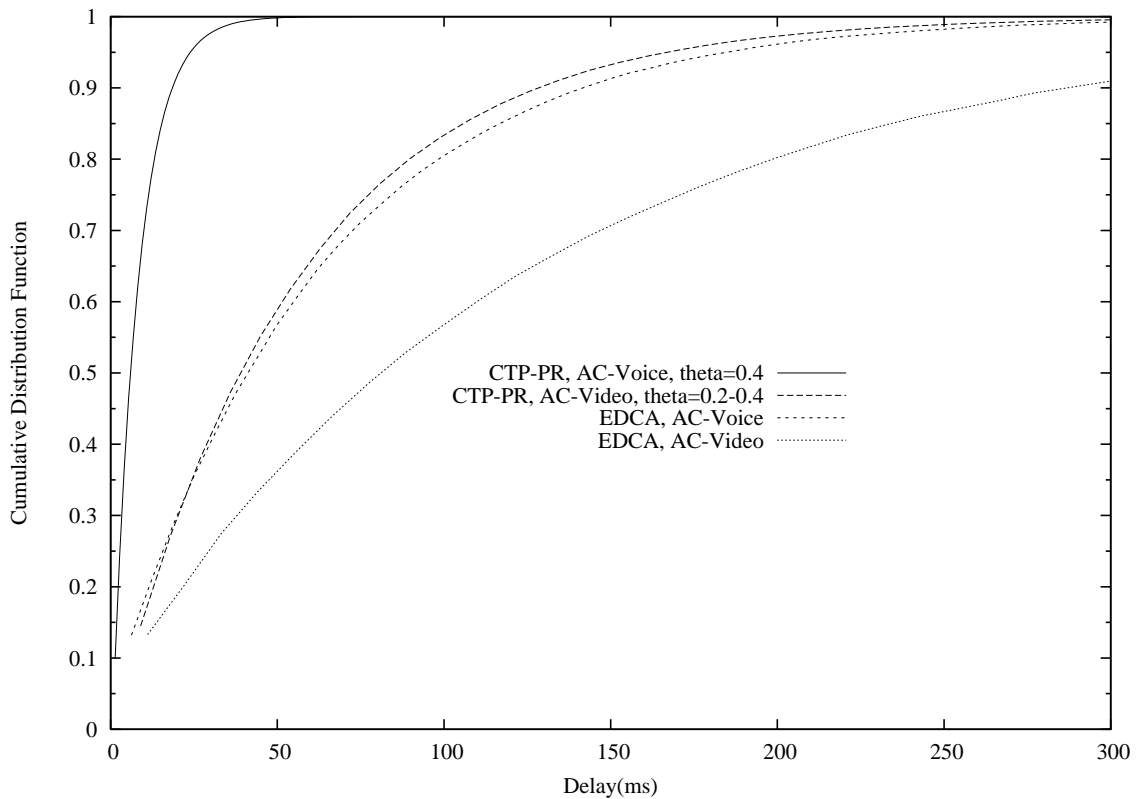


Figure 3.12: Simulation results for delay distribution.

The delay measurements based on Eqn. (9) are depicted in Fig. 3.11 which compares the saturation throughput performance of CTP with QoS, which is referred to as CTP-PR, to that of EDCA. The comparison shows significant improvement in delay performance in all ACs. The author further shows the simulation results for the delay distributions in Fig. 3.12 for AC3 and AC2 that carry delay sensitive traffic. The results show that CTP-PR provides over 90% one-hop delivery delay within 20 ms for AC3 traffic whereas EDCA struggles to reach 50% for the same delay requirement. Likewise, CTP-PR delivers over 80% of video traffic within 100 ms while EDCA manages only 50%.

It was mentioned earlier that one of the primary advantages of services differentiation using CTP is the low collision probability, which allows such a design to scale with increasing node density. The collision probability of CTP-PR is compared with that of EDCA in Fig. 3.13. The collision probability for CTP-PR stays below 2% at

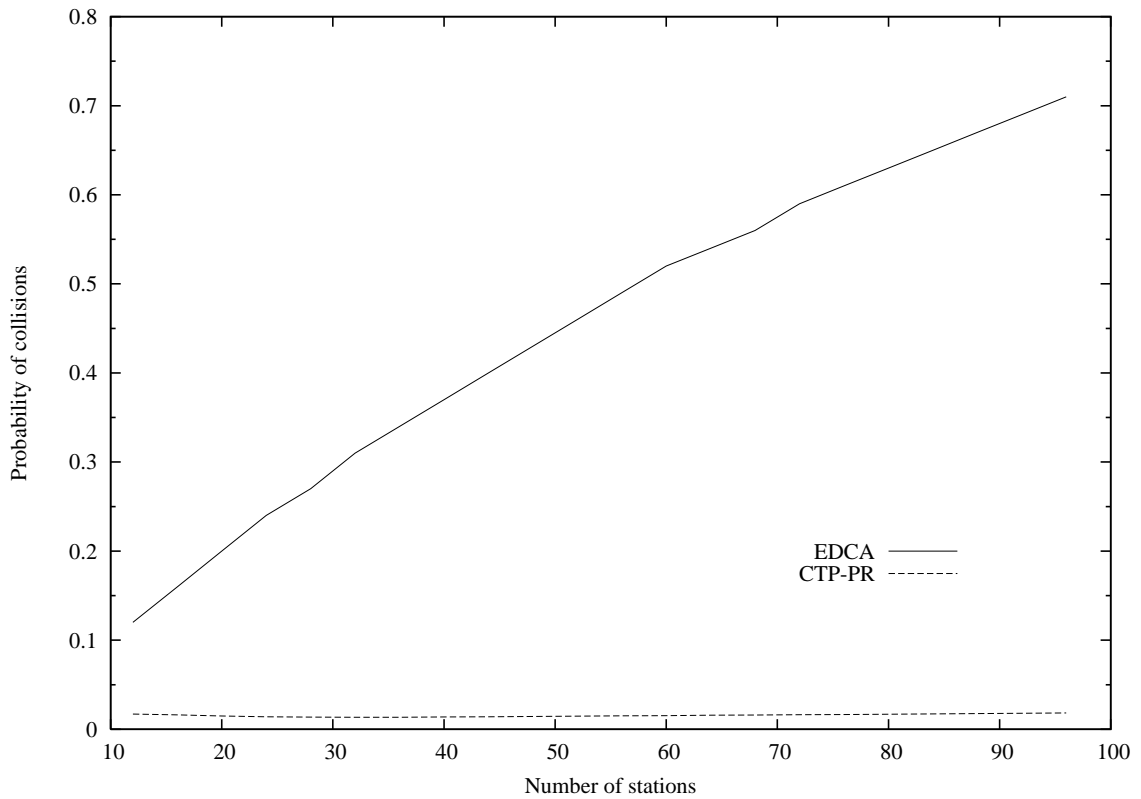


Figure 3.13: Simulation results for collision probability of service differentiation using CTP when compared to EDCA.

all densities, thereby allowing for high channel utilization.

Next, the author validates the effectiveness of the proposed design for eliminating the access point (AP) bottleneck problem. The setup consists of a scenario where there is one AP for a set of stations with equal number of stations for each access category. In Fig. 3.14, the author shows that the proposed design provides higher throughput for AP to cope with the downlink traffic. In Fig. 3.15, the author compares performance of the system with and without giving the AP additional chances based on analytical results. In the latter case, where the AP contends with the same priority as other voice stations, stations with voice data can be seen to enjoy a success rate of 75% or above. However, the effective success rate is confined to around 10% or less due to the communication bottleneck at the AP. As the AP contends like any of the other voice stations, it results in a communication bottleneck for all other voice data as all of them are routed through a single AP. Eliminating this bottleneck

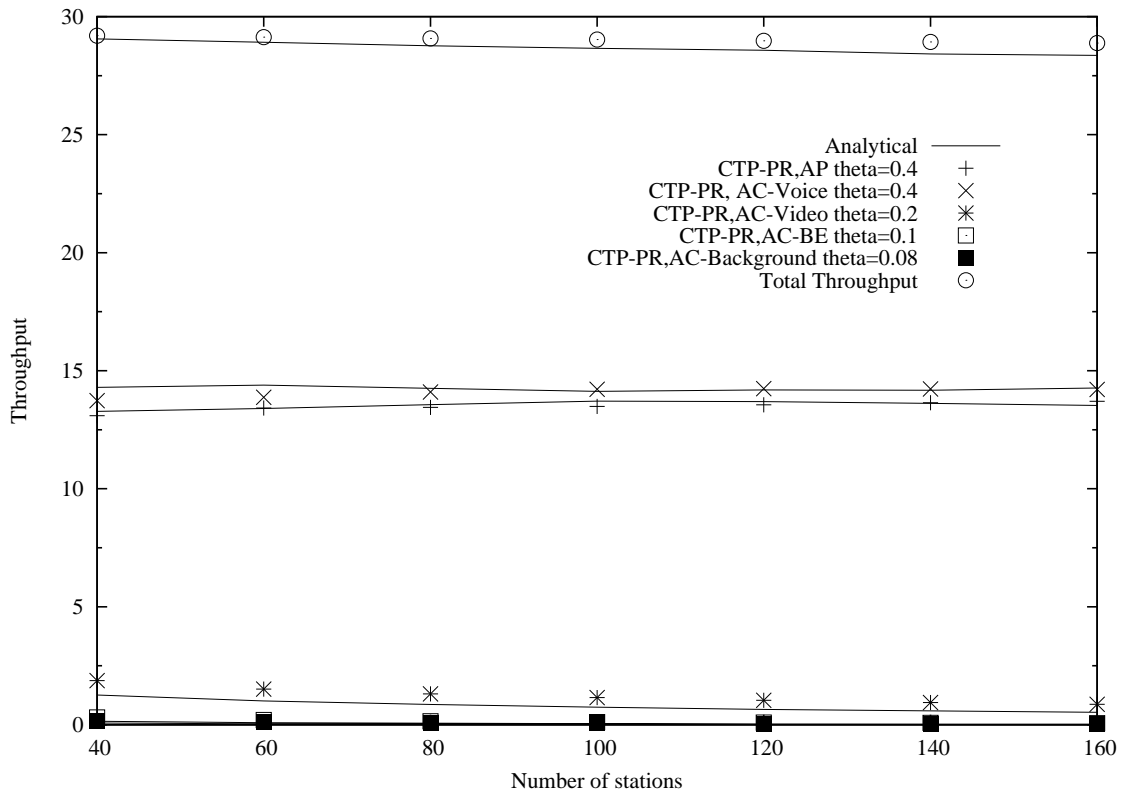


Figure 3.14: Throughput vs the number of stations obtained from simulations.

by giving the AP additional chances increases the support to about 45% of the total bandwidth. It is further observed that the total throughput presented maintains close to the maximum achievable throughput. This shows that the introduction of the proposed method results in no negative impact to the performance.

3.4 Discussions and Future Work

Based on the analytical and simulation results, the author can conclude that implementation of the proposed mechanism can help realize effective service differentiation as evidenced by the high throughput and low delays. In this section, the author discusses other aspects of protocol behaviour which are not the focus of this chapter. Instead, the author makes note of how different features of wireless network operation can impact the protocol performance and thus form guidelines for future study.

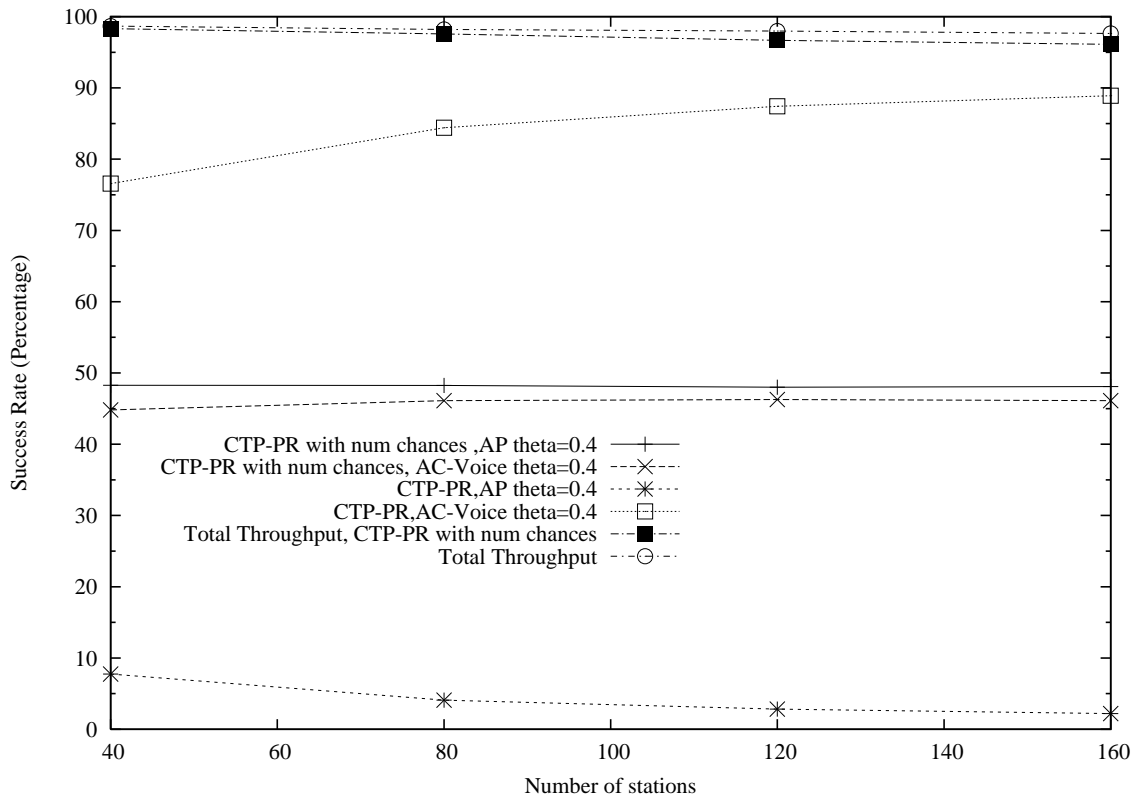


Figure 3.15: Analytical performance with and without giving chances to the AP.

As mentioned earlier, saturated network conditions have been considered in the simulations so as to identify how differentiated channel access maps to the actual data transmission performance. However, in a practical scenario, real-time traffic such as voice and video are unlikely to be saturated because of their application characteristics. Therefore, additional experiments using realistic packet arrival models have been conducted to supplement the simulation results. The author uses VoIP and MPEG-4 streams based on models used in existing literature [49, 92, 93]. ON-OFF sources are used for VoIP and CBR flows for MPEG-4 traffic. Packet sizes were also chosen accordingly. The packet generation for low priority traffic (AC0 and AC1) is kept the same as before, with saturated arrivals. Subsequently, the delay performance for AC3 and AC2 are obtained. It is important to note here that since the packet sizes are shorter and follow non-saturated arrival for these streams, the mean throughput cannot be considered to be a true measure of performance since it gets normalized.

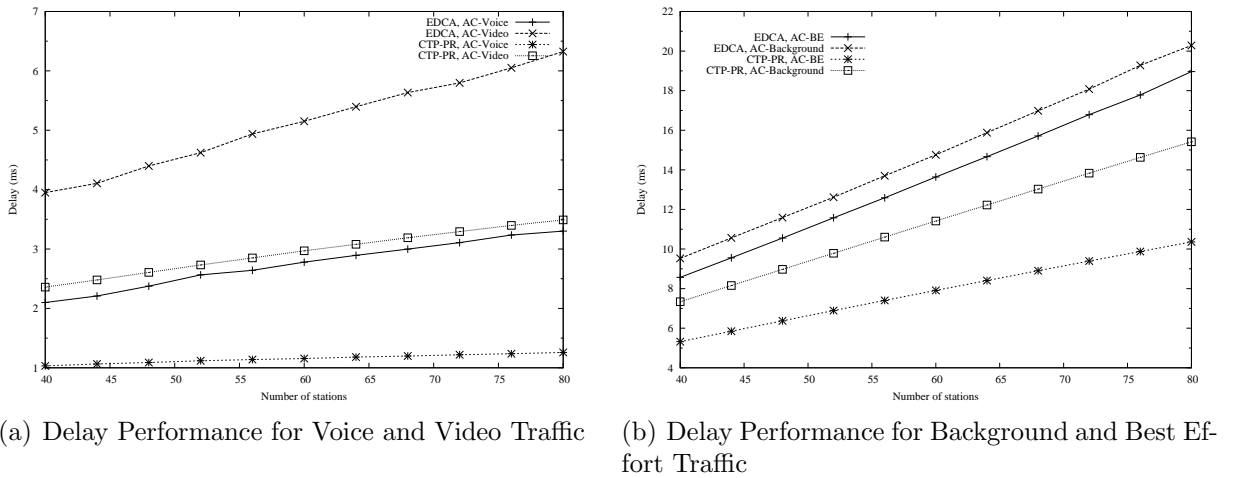


Figure 3.16: Mean Delay performance using realistic models for VoIP and MPEG-4 streams.

The delay performance is compared to that of EDCA in Fig. 3.16. As can be seen, the results follow the same trend as that of saturated traffic even though the exact values are smaller because of shorter transmission times.

Apart from a deterministic delay, the QoS requirements for real-time traffic are typically an ensemble of a larger set of parameters. As part of the IEEE 802.11e standard [6], these have been defined in the form of a set of traffic specifications (TSPEC) which is used by the access point for admission control. As discussed earlier, the channel access uses a separate set of parameters. Since the design of parameters for the proposed contention tone based mechanism uses the latter as a reference, admission control can proceed in the same manner as in 802.11e. An expected benefit with regard to this aspect of performance is the admission of more streams with strict QoS requirements. In addition, from the perspective of the proposed design of device differentiation in this chapter, the author foresees the scope of enhancing admission control by balancing the parameter settings for the AP with that of the streams being admitted. However, since admission control in itself constitutes a critical aspect of the QoS performance of a network, any design that influences its operation needs to be viewed in its context. For instance, in this case, the contention tone behaviour needs to take into consideration the various QoS parameters specified in the TSPEC. A

future direction for this work is, therefore, to extend the benefits observed in channel access to other aspects of QoS design.

A second assumption made for the simulations is that all nodes transmit using the same constant data rate. However, in a practical network scenario, nodes are likely to choose a data rate at which to transmit based on existing channel conditions or transmission distance. The author does not expect this in itself to affect the basic functionality of the protocol since the channel contention is independent of the rate of data transmission. An implicit requirement is that all nodes receive any contention tone transmitted when they are not transmitting themselves. Since this only involves detection of a transmission burst, rather than receiving a packet, it is fair to assume that all nodes within the transmission range can also access the contention-tone channel. However, a more critical design aspect which could be impacted due to the presence of multiple data rates is the length of contention segments. In case of transmission at very high data rates, the data transmission itself may be shorter than the contention period, resulting in idle time slots spent in determining the winner. On the other hand, increasing the channel bandwidth can allow for faster detection and therefore, shorter contention period. However, these tradeoffs would need to be balanced with those outlined earlier in Section 3.1. Hence, a comprehensive design for adapting the contention tone protocol to the multi-rate scenario needs to be based on a detailed study of the tradeoffs involved. In the simulations for this chapter itself, the highest available data rate is used and it does not give rise to the above mentioned issue which thereby excludes the probability of the issue taking place at lower data rates.

While the issue of hidden terminals is not dealt with as part of the current discussion on CTP, occurrences of the problem are expected to be limited due to the high success rate of the proposed design. Nevertheless, additional design choices can still be explored. As with the existing IEEE 802.11 DCF, RTS/CTS can be included as part of packet transmission on the data channel, though this is likely to reduce the overall throughput somewhat. An alternative methodology is a design based on the

transmission range of the CT transmission burst. A larger range for CT than that of data transmission would imply fewer occurrences of the hidden terminal problem. For both cases, as with the problem of multi-rate transmission, the impact of the length of contention segments and the CT channel bandwidth would need to be looked into.

3.5 Conclusion

Multi-channel MAC protocols have shown potential in improving the efficiency of WLANs. There are also obvious reasons to include the design of service differentiation in the potential multi-channel MAC protocols. In this chapter, the author proposes a service and device differentiation scheme for contention-tone based WLAN protocol. In the proposed design, stations contend for channel access by varying their CT transmission probabilities and making use of multiple chances to compete in a CT contention segment. While an appropriate choice of CT transmission probabilities alone could suitably achieve service differentiation, the design choice of multiple chances is used to allow the AP to schedule downlink voice streams more effectively. Thus, the scheme addresses the AP bottleneck problem which is not handled in many QoS differentiation mechanisms for WLANs, including the IEEE 802.11e. The author models the competition and studies the probability of successful channel assignment. Presented numerical results have confirmed the effectiveness of the service and device differentiation, and the efficiency of the proposed scheme.

Along with the analysis of the proposed scheme, the author also presents simulation results which not only validate the developed model, but also give insight to the delay performance of the proposed scheme.

Chapter 4

Stateless Network-wide Broadcast - Analysis and Design

The contributions of this chapter are divided into two sections. The first part is devoted to a comprehensive analysis of stateless broadcasting to understand the benefits of implicitly available *network information*. The second part focuses on analysis and design of stateless algorithms for multi-rate broadcasting.

4.1 Performance Improvements for Network-Wide Broadcast with Instantaneous Network Information

The focus of this section is to analyze the effect of implicitly available *network information* on network-wide broadcasting in wireless ad hoc networks. The primary motivation for this is that the resulting performance is independent of any transmission overheads. Furthermore, the instantaneous nature of *network information* implies that algorithm design can adapt to dynamic network topologies. An obvious choice for analysis of the impact of *network information* on network-wide broadcast is to study stateless broadcasting algorithms. The author characterizes the *network*

information available at a node during different phases of broadcasting. This allows the author to concentrate the discussion on a generalized view of stateless broadcasting rather than individual algorithms. Subsequently, the author develops a unified analytical model to evaluate broadcasting performance. To the best of the author's knowledge, the proposed work is the first work that provides such a generalized approach for stateless broadcasting performance analysis.

The author first studies the performance under ideal channel conditions using a unit disk graph (UDG) model of a network. The author analytically studies how forwarding probability of nodes is impacted by the *network information* at each stage of broadcasting. The author's analysis methodology looks at how broadcasting performance is affected as the message propagates through the network. The author also analyzes the redundancy of transmissions and how they compare with the conclusions drawn by Ni et al [8]. The analytical results are validated with the simulation results. The simulation setup involves broadcasting as part of route discovery, which allows the evaluation of other aspects of the protocol performance. The key difference of the author's work with existing analytical studies is that, instead of analyzing individual schemes, the author clearly shows the impact of *network information* on the performance.

Subsequently, the author uses the analytical results to draw insights on the performance of broadcasting algorithms in general and discusses how the reliability of broadcasting is affected by the node density in the network. In addition, the author obtains feasibility regions within which a broadcast algorithm needs to operate given the network costs it incurs. Lastly, the author adapts the proposed model into a realistic approximation of the network conditions by using a Quasi-Unit Disk Graph (QUDG) model to replace the UDG model. This facilitates the analysis of broadcasting reliability in dynamic channel conditions and subsequently allows the author to draw insights for robust algorithm design.

4.1.1 Statement of Objectives

The salient features of stateless broadcasting are identified which subsequently act as guidelines for the discussion in the rest of section 4.1. The performance objectives of any broadcast algorithm can be summarized as

$$\begin{aligned} & \min n_{tx} \times e_{avg} \\ & s.t. \sum_{i=1}^N r(i) = N \end{aligned} \quad (4.1)$$

where n_{tx} denotes the total number of broadcasts in the network while e_{avg} is the average energy consumption for a data transmission. The term e_{avg} is important it is used as a measure of the energy consumption involved as part of the algorithm design. Thus, this accounts for additional overheads incurred by the broadcasting algorithm. The second part of (4.1) identifies the objective of network coverage. For a node i in a network with N nodes, $r(i)$ takes the value 1 when the node receives the broadcast and 0 otherwise.

The objectives outlined above can easily be mapped to the objective of addressing the *Broadcast Storm* problem. As with existing protocol designs and from the results in [2], it is safe to assume that minimizing the number of transmitting nodes (n_{tx}) results in minimizing the contention and collision overheads. A combination of both the conditions intuitively implies minimizing the redundancy while the parameter e_{avg} explicitly takes care of the energy costs.

The analysis methodology followed in the rest of the section is based on modeling the fraction of forwarding nodes required for broadcasting. Subsequently, this model is used to obtain design insights from the point of view of ensuring reliability of broadcasting and the impact of protocol overheads (in terms of the resulting energy consumption).

4.1.2 Network Information

The author has earlier introduced *network information* as the information derived from overheard transmissions in the neighbourhood of a node. In the context of stateless broadcasting, this needs to be mapped to the transmission effectiveness of the node in terms of the expected additional coverage (EAC). The EAC of a forwarding node v is the area of the region which first receives the broadcast packet from v . Thus, the transmission effectiveness of a forwarding node increases with the EAC. For distance based broadcasting schemes, the distance from the transmitting node is used to infer the EAC. In counter based schemes, the number of transmissions overheard is used as a measure. For protocol design using both schemes, a random or deterministic delay is introduced subsequent to a node receiving the first copy of the broadcast message. This is to increase the accuracy of the estimated EAC before it decides on its transmission behaviour. Using these salient features of stateless algorithms, the author distinguishes the *network information* available at a node into two distinct phases. As the delay used in both schemes is protocol specific, the author does not focus on it as an aspect of *network information*. Instead, the effect of transmissions in a node's neighbourhood while it waits for channel access is identified.

As part of recognizing the broadcast from a neighbouring node, the node can estimate the effectiveness of its transmission if it chooses to take part in forwarding. This is termed the *broadcast recognition phase (BRP)*. Thus, the *network information* available can be expressed in terms of the distance to the transmitting node. A distance threshold denoted by a is defined as a fraction of the transmission range. Given a uniform distribution of nodes, effectiveness of a node's transmission is maximized if it lies at the periphery of the transmission range. The author considers a behaviour in which nodes receiving the first copy of message from a node within the range a choose not to forward. The dependence of the forwarding probability of nodes as a function of a is examined in the analysis.

In addition to the *network information* obtained from the first phase, a node's transmission effectiveness also depends on the number of broadcasts in its neighbour-

hood, which forms a basis for counter based schemes [8]. This is termed the *broadcast optimization phase (BOP)*. Each overheard broadcast results in a reduction of the EAC at a node and therefore, the effectiveness of its transmission. Furthermore, this dependence grows stronger for broadcasts received over shorter distances. Based on this, the author considers a behaviour wherein nodes probabilistically decide on forwarding behaviour upon overhearing broadcasts from neighbours located nearer than a threshold distance, d_{th} . The packet discarding probability, p_{disc} is also a function of the distance threshold, and is discussed in the next section.

Here, it is interesting to note how these parameters map to protocol specific behaviour of the existing schemes. Firstly, existing mechanisms make use of a random delay to decide on their forwarding behaviour. Overheard broadcasts during this duration impact the node's forwarding behaviour which is determined at the end of the delay. By allowing the forwarding behaviour to be determined purely in terms of neighbourhood transmissions, the protocol specific operations can be mapped to the actual MAC layer ones. Furthermore, the forwarding behaviour reduces to that of a basic distance based scheme when $d_{th} = a$ and $p_{disc} = 1$.

Defining the parameters based on the distance between nodes also allows the author to stay true to the broadcast performance objective of identifying a CDS in the network. For the unit disk graph realization of a wireless network, construction of a CDS implies maximizing the distance between them [94]. Furthermore, maximizing the distance is also necessary for optimal scheduling among broadcasting nodes [95].

The analysis in the rest of the chapter is based on *network information* as conceptualized above. Here, it is noted that there are likely to be scenarios in which the *network information* at nodes is inaccurate. The current discussion focuses on identifying the maximum benefits achievable using *network information*. Inaccurate *network information*, therefore, is likely to result in inferior performance. For instance, inaccurate estimation of the distance between nodes is likely to result in a higher percentage of nodes acting as forwarding nodes. Since identification of the precise effect of such inaccuracies depends on individual scenarios, such issues are

expected to be handled as part of specific protocol definitions. For instance, use of an accurate delay can help adapt to topology changes. The impact of inaccurate estimation on multi-rate broadcasting schemes has been discussed in section 4.2.2.3. An additional consideration is the acquisition of the *network information* itself. As per the motivation, this is expected to incur zero or minimal overheads. In ad hoc network deployments such as sensor networks or even scenarios in which devices use similar network interfaces, all nodes can be expected to use constant transmission power. Therefore, the distance estimation can be done based on the received signal strength to a high degree of accuracy. In more dynamic topologies, only the source transmission power may need to be piggybacked incurring a small amount of overhead.

4.1.3 Broadcast Performance Analysis Under Ideal Channel Conditions

4.1.3.1 System Model

The author is motivated to understand the impact of *network information* at each of the two phases on the broadcast performance. It is necessary to map node transmission behaviour to the estimated effectiveness of the transmission based on the available *network information*. Node transmission behaviour is defined from the perspective of available *network information*. The network model considered is that of a Unit Disk Graph (UDG) [96] in which all nodes have a fixed transmission radius of unit distance.

In the *broadcast recognition phase*, only nodes lying outside the range characterized by the distance threshold, a , take part in forwarding since all other transmissions are estimated to be ineffective. Similarly, in the *broadcast optimization phase*, the transmission effectiveness reduces probabilistically with the number of broadcasts heard over a close range. Therefore, the author defines a behaviour in which nodes probabilistically decide not to forward upon hearing a rebroadcast from a node lying closer

than a threshold d_{th} . For ease of discussion, the author uses $d_{th} = a$. In order to ensure that the resulting behaviour closely maps to the effectiveness of the transmissions, p_{disc} is defined on the basis of a set of observations. As mentioned earlier, the effectiveness of a transmission with respect to an overheard broadcast would depend on the distance between the two nodes. Specifically, a node's transmission is likely to be redundant if it is located very close to the broadcasting node. Thus, a smaller value d_{th} would imply that every overheard broadcast reduces the effectiveness at a faster rate. The converse would be true for larger values of d_{th} . The same would be true for different broadcasts received from within the range of d_{th} , given a fixed value for d_{th} . Averaging the effect of all transmissions from within the range of d_{th} and using $d_{th} = a$, p_{disc} is defined as,

$$p_{disc} = \left(1 - \frac{a}{2}\right) \quad (4.2)$$

Considering a fixed unit transmission radius, $0 < a < 1$. To effectively understand the interplay between transmission behaviours of nodes, the author considers the entire set of nodes as divided into regions of concentric circles marked by their transmission range, with the source at the centre. These regions are called hop ranges. Thus, a node that requires at least one forwarding node, and therefore two transmissions, to reach the source lies in the second hop range. Transmission coverage region is denoted as TR . The region lying within the distance threshold is referred to as the alerted region and is denoted by AR . Forwarding nodes are likely to be located in the region between TR and AR , which is denoted as FR . Since the author uses $d_{th} = a$, AR is also used to denote the range marked by the distance threshold for *broadcast optimization phase*. In the rest of the discussion, MSG is used to denote the message being broadcast in the network.

4.1.3.2 Broadcast recognition phase

The author first identifies the effect of the available *network information* during the *broadcast recognition phase* on the forwarding probability of nodes. Since this does

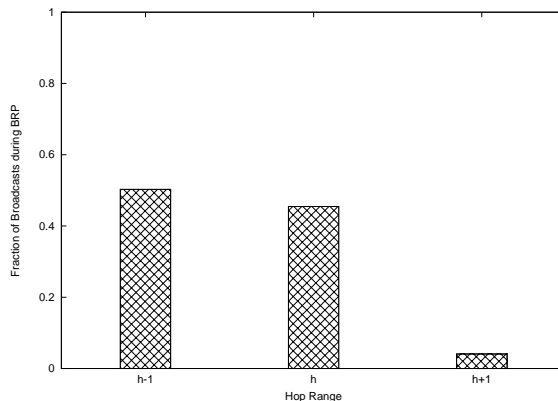


Figure 4.1: Fraction of broadcasts heard by a node in hop range h during BRP .

not include the effect of the *broadcast optimization phase*, the forwarding behaviour of nodes would only depend on the first copy of MSG received. The discussion hereafter concerns the probability that a node in a hop range h either rebroadcasts an overheard packet or refrains from doing so. These two probabilities are denoted by $p_t(h)$ and $p_r(h)$ respectively. Uniform distribution of nodes in the network is assumed.

The author starts by obtaining the transmission probabilities in the first hop range and subsequently obtain generalized expressions for the succeeding hop ranges. The analysis for $h = 1$ is straightforward and on the lines of analysis of existing distance based mechanisms [60, 53]. All nodes lying within the AR would refrain from forwarding while those lying in FR would transmit. Therefore, the forwarding probability values for the first hop range would be obtained as,

$$\begin{aligned} p_t(1) &= 1 - a^2 \\ p_r(1) &= a^2 \end{aligned} \tag{4.3}$$

Next the transmission behaviour for nodes in $h > 1$ is examined. Since the nodes are not within the transmission range of the source, the transmission behaviour of a node is determined by the distance to the node from which it receives the first copy of MSG . This can correspond to one of two different scenarios which can impact the behaviour. In the first case, a node in h overhears MSG from a node lying in

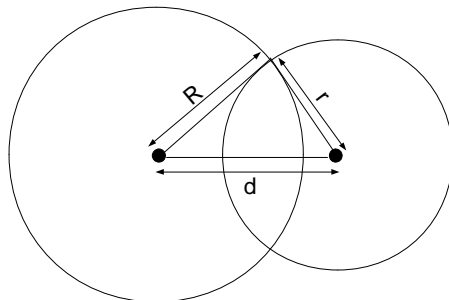


Figure 4.2: Overlapping area between two circles of radii R and r with centers separated by a distance d .

$(h - 1)$, which is termed *Scenario 1*. Alternatively, a node in h could receive *MSG* from another node in h , which is called *Scenario 2*. By design, the broadcasting node in *Scenario 2* would have heard *MSG* from a node in $(h - 1)$ and lies in FR . A third scenario is that the transmission of *MSG* from $(h + 1)$ impacts transmission in h . However, the likelihood of this occurring is low since the node in h is likely to have overheard at least one of the preceding transmissions from $(h - 1)$ and h . This is confirmed by simulations in which the probabilities of a node in hop range h receiving the first copy of *MSG* from $(h - 1)$, h and $(h + 1)$ are measured. The results are shown in Fig. 4.1. Hence, this scenario is not included in the derivations. As a result of the scenarios mentioned above, nodes that transmit and those that do not are unlikely to be distributed according to a specific geometric pattern for $h \geq 2$ unlike $h = 1$. The analysis hereafter is focused on identifying the forwarding probability as a result of *Scenario 1* and *Scenario 2*. These probability values are then used to obtain the fraction of nodes in the network that transmit.

As node broadcasting behaviour depends on the relative locations of the nodes with respect to each other, the overlapping regions between transmission ranges of nodes need to be considered, which can be obtained from circle geometry. The area of intersection between any two circles of radii r and R with the centres located at a

distance d from each other can be given as,

$$\begin{aligned}
 A(d, r, R) = & r^2 \cos^{-1}\left(\frac{d^2 + r^2 - R^2}{2dr}\right) + R^2 \cos^{-1}\left(\frac{d^2 + R^2 - r^2}{2dR}\right) \\
 & - \frac{1}{2} \sqrt{(-d + r + R)(d + r - R)(d - r + R)(d + r + R)} \quad (4.4)
 \end{aligned}$$

The overlapping area between two intersecting circles is illustrated in Fig. 4.2.

Scenario 1: For any node in hop range h , the set of transmissions accounting for *Scenario 1* lies in the overlapping area of its transmission range and region of $(h - 1)$. The latter is the circle with radius $(h - 1)$ (since transmission range is normalized to 1) with the source at its centre. The set of transmissions of *MSG* that constitutes *Scenario 1* lies in the area of intersection of *AR* and the circular region for $(h - 1)$. Thus, for any node located at a distance x from the source and lying in the hop range h , the probability that the copy *MSG* it receives first is from the hop range $(h - 1)$ and refrains from broadcasting the overheard *MSG* can be given as

$$p_r(h, h - 1, x) = p_t(h - 1) \frac{A(x, a, h - 1)}{A(x, 1, h - 1)}. \quad (4.5)$$

It is important to clarify the author's use of notation here, so as to understand the difference between $p_r(h)$ and $p_r(h, h', x)$. The latter notation $p_r(h, h', x)$, as used above, is the probability that any node, located at a certain distance x from the source, refrains from broadcasting upon overhearing the first copy of *MSG* from a node in h' . This will be used later to obtain the probability that any node in hop range h refrains from transmission, given as $p_r(h)$.

Scenario 2 (S2): Similar to *Scenario 1*, the probability that a node decides not to forward *MSG* in this case would depend on the size of the overlapping region between the hop range h and the *AR* for the node. However, this probability would also indirectly depend on the transmissions taking place in $(h - 1)$. This is referred to as *Phenomenon-S2* and will be elaborated here. For visualization of the way *MSG* is propagated across the network, the first copy to enter hop range h must have been transmitted by some node in $(h - 1)$. Subsequently, the set of nodes in h ,

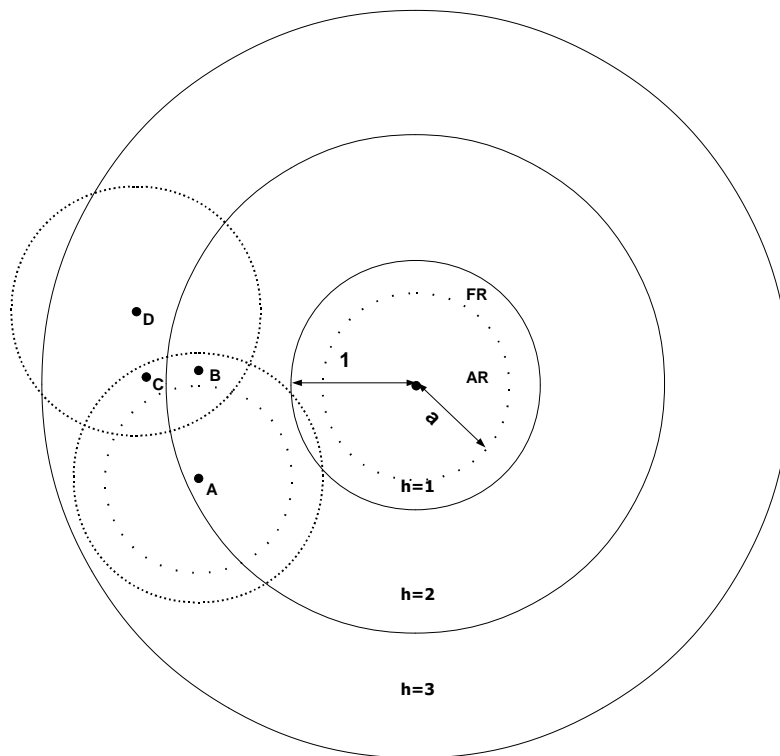


Figure 4.3: Illustration on the distribution of forwarding nodes.

say T_h , which are located in the FR s of the transmitted copies of MSG contend to broadcast their copies. If one of the nodes in T_h transmit in the succeeding time slot (i.e. immediately after having received MSG from $(h - 1)$), the set of nodes whose transmission behaviour it impacts would only be those which are not in the range of the previous transmission of MSG (from $(h - 1)$). The nodes impacted by a later transmission would similarly be those that are outside the range of all the preceding transmissions. Here, it is important to note that the set of nodes that T_h contends with includes those in $(h - 1)$ that have yet to transmit. Thus, the probability that the first copy of MSG a node in h receives, is from a node in h , is the probability that it has not overheard any of the preceding broadcasts from $(h - 1)$.

To better understand *Phenomenon-S2*, refer to Fig. 4.3 in which nodes A and B are located in the second hop range while nodes C and D are in the third, S being the source. Subsequent to the source having transmitted MSG , fast forward to the state in which all nodes in the second hop range have received MSG . Assume that both A

and B take part in broadcasting on account of being located in FR when receiving their first copies of MSG . Suppose A first gains access to the channel with only C overhearing its transmission in the third hop range. Since C does not lie within A 's AR , it attempts to rebroadcast and contends for channel access with B , which has yet to transmit. Therefore, the probability that D 's transmission behaviour is determined by C depends on the probability that the latter transmits before B does.

Thus, the analytical model for *Scenario 2* needs to include *Phenomenon-S2*. As discussed above, this is the probability that a node i in h overhears MSG from another node in h before one from $(h - 1)$. This would depend on how channel contention proceeds among i 's neighbours. Now, the channel access probability for any node would depend on the size of this neighbourhood. From Fig. 4.3, it can be concluded that the transmission probability of a node in $(h - 1)$ would reduce as more nodes in h start contending upon having received a copy of MSG . However, the exact probability of *Phenomenon-S2*, i.e. the impact of nodes in $(h - 1)$ on *Scenario 2*, is difficult to be estimated geometrically since it depends on the contention behavior of nodes. Nevertheless, such behavior can still be approximated based on the fact that the number of nodes contending grows in proportion to the area of the FR . Therefore, it can be concluded that the probability that a node in $(h - 1)$ transmits and thereby impacts hop range h , scales with $(1 - a^2)$ on average. Thus, the probability that a transmission from $(h - 1)$ does not reduce the effectiveness of one in h can be approximated as,

$$p_c = [1 - (1 - a^2)p_t(h - 1)] \quad (4.6)$$

It is worth noting here that a similar phenomenon is likely to take place for Eqn. (4.5) where transmission from the hop range h could influence the effectiveness of transmissions from $(h - 1)$. However, the probability of the same is expected to be relatively small since any node in h would have received MSG either directly or indirectly from $(h - 1)$. For any other node still contending in $(h - 1)$, the additional coverage area in h would have been reduced by other transmissions from $(h - 1)$. Hence, the effect of a subsequent transmission from h would be negligible.

Finally, the probability that a node in h refrains from forwarding in *Scenario 2* is obtained. This would depend on the probability that a copy of MSG is overheard from a node located in hop range h and within the AR of the receiving node. For a node located at a distance x from the source, this area can be given as $(A(x, a, h) - A(x, a, h - 1))$. Thereafter, the probability that such a transmission is not impacted by other transmissions in $(h - 1)$ is accounted for as per equation (4.6). Thus, the probability that a node in h refrains from forwarding in *Scenario 2* is obtained as,

$$p_r(h, h, x) = [1 - (1 - a^2)p_t(h - 1)]p_t(h) \frac{(A(x, a, h) - A(x, a, h - 1))}{(A(x, 1, h) - A(x, 1, h - 1))} \quad (4.7)$$

The probability that a node refrains from forwarding is the probability that it is located in AR when receiving the first copy of MSG , in either *Scenario 1* or *Scenario 2*. Now, a node situated at a distance x from the source can lie anywhere on the circle with circumference $2\pi x$ while the region covered in the hop range h is a concentric circle with area $(2h - 1)\pi$. Thus, the probability that any node in the hop range h refrains from forwarding MSG can be expressed as

$$\begin{aligned} p_r(h) &= \int_{h-1}^h p_r(h, h - 1, x) \frac{2x}{2h - 1} dx + \int_{h-1}^h p_r(h, h, x) \frac{2x}{2h - 1} dx \\ &= \int_{h-1}^h p_t(h - 1) \frac{A(x, a, h - 1)}{A(x, 1, h - 1)} \frac{2x}{2h - 1} dx \\ &\quad + \int_{h-1}^h [1 - (1 - a^2)p_t(h - 1)]p_t(h) \left(\frac{A(x, a, h) - A(x, a, h - 1)}{A(x, 1, h) - A(x, 1, h - 1)} \right) \left(\frac{2x}{2h - 1} \right) dx \end{aligned} \quad (4.8)$$

The probability that any node in hop range h transmits is that it does not refrain from doing so and can be obtained as $p_t(h) = 1 - p_r(h)$. Solving the implicit linear equation, $p_t(h)$ can be expressed in terms of the transmission probability of the previous hop range as,

$$p_t(h) = \left(\frac{1 - \int_{h-1}^h p_t(h - 1) \frac{A(x, a, h - 1)}{A(x, 1, h - 1)} \frac{2x}{2h - 1} dx}{1 + \int_{h-1}^h [1 - (1 - a^2)p_t(h - 1)] \frac{(A(x, a, h) - A(x, a, h - 1))}{(A(x, 1, h) - A(x, 1, h - 1))} \frac{2x}{2h - 1} dx} \right) \quad (4.9)$$

Using Eqn. (4.3) in Eqn. (4.9) and subsequently in Eqn. (4.8), the author obtains the probability values for the second hop range and in turn use the values for succeeding hop ranges.

Given the node transmission probabilities for successive hop ranges, the fraction of all nodes that take part in route discovery can be obtained by summing the fraction of nodes transmitting in each hop range. As discussed before, the first hop range is the circular area covered by the transmission range of the source while each subsequent hop range is a concentric circle with increasing area. The fraction of transmitting nodes can thereby be obtained as,

$$F_{tx} = \frac{p_t(1)\pi + \sum_{i=2}^H p_t(h)(2h-1)\pi}{\pi H^2} \quad (4.10)$$

where H is the maximum number of hop ranges which would depend on the network topology. In the simulation scenario, the source s is located at the centre while the destination d is located at the edge of the circular simulation region. Therefore, the value of H can be obtained as $H = \frac{dist(s,d)}{TX_RANGE}$, where $dist(s,d)$ is the distance between the source and the destination while TX_RANGE is the constant transmission range for any node in the network. The area of the total simulation region is, therefore, πH^2 .

4.1.3.3 Broadcast Optimization Phase

Next, the author evaluates the additional performance benefits achievable by utilizing *network information* available during the *broadcast optimization phase*. The resulting forwarding probability of nodes would depend on the discarding probability p_{disc} in addition to the value of a . Here, it is observed that any node that can utilize the *network information* available in the *broadcast optimization phase* is one of the potential forwarding nodes after the *broadcast recognition phase*. For any particular hop range h , this is the same as the transmission probability obtained in Eqn. (4.9), given as $p_t(h)$. Thus, the probability that a node refrains from forwarding as a result

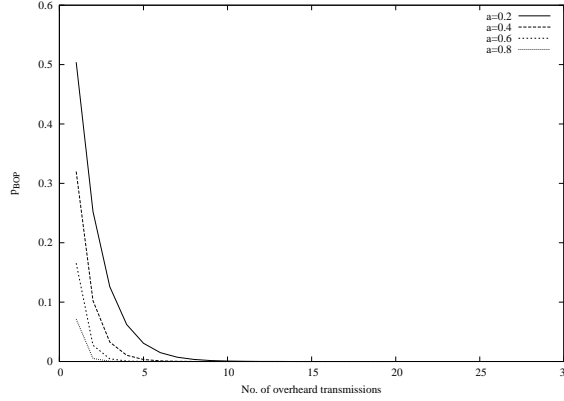


Figure 4.4: Figure showing probabilities of retransmission.

of the *network information* available in the *broadcast optimization phase* follows from Eqn. (4.8) as

$$\begin{aligned}
 p_{rd}(h) = p_{disc}p_{ts}(h) & \left[\int_{h-1}^h p_t(h-1) \frac{A(x, a, h-1)}{A(x, 1, h-1)} \frac{2x}{2h-1} dx \right. \\
 & \left. + \int_{h-1}^h p_{ts}(h) \frac{(A(x, a, h) - A(x, a, h-1))}{(A(x, 1, h) - A(x, 1, h-1))} \frac{2x}{2h-1} dx \right]. \quad (4.11)
 \end{aligned}$$

Thus, the probability that any node refrains from forwarding after all the phases can be expressed as

$$p_{rs}(h) = p_r(h) + p_{rd}(h) \quad (4.12)$$

The corresponding transmission probability can be obtained as $p_{ts}(h) = 1 - p_{rs}(h)$. The fraction of nodes that transmits in the entire network can be obtained by substituting $p_{ts}(h)$ for $p_t(h)$ in Eqn. (4.10).

4.1.3.4 Effect on Redundancy

Having obtained the forwarding probability of nodes in stateless broadcasting, the author is interested in analyzing the effect on the redundancy of transmissions. As discussed before, a node's transmission probability needs to be closely mapped to its effectiveness, which is in terms of the EAC [8].

For the *broadcast recognition phase*, the effect of *network information* on redun-

dancy is straightforward. Since node transmission probabilities only depend on the first copy of MSG received, the effect is deterministic. The only reduction in redundancy is due to the fraction of nodes that do not transmit on account of having being located in the AR upon receiving the first copy of MSG .

In the case of the *broadcast optimization phase*, the author identifies how Eqn. (4.2) translates into reducing the number of redundant transmissions. The probability of a node discarding its packet upon overhearing multiple broadcasts of MSG is derived. As observed in [8], the probability should reduce exponentially in order to minimize redundancy. The corresponding probability for any node within the range of the source is identified. Since the author analyzes the effects on redundancy as a result of the *broadcast optimization phase*, the nodes under consideration lie in the FR . p_{AR} and p_{FR} denote the probabilities that a node lies in AR or FR respectively for any subsequent broadcast of MSG that it overhears. The probability that a node transmits upon hearing a broadcast from within AR can be given as $p_{AR}(1 - p_{disc})$. Hence, the probability that a node transmits after having overheard n broadcasts in the *broadcast optimization phase (BOP)* can be given as,

$$p_{BOP}(n) = \sum_{k=0}^n \binom{n}{k} [p_{AR}(1 - p_{disc})]^k p_{FR}^{(n-k)} \quad (4.13)$$

The corresponding values are plotted in Fig. 4.4. Note that the value of n does not include the first copy of MSG received as part of the *broadcast recognition phase*. As the value of a ¹ increases, the forwarding probability of a node drops sharply with the number of overheard broadcasts, which implies fewer redundant transmissions. A higher value of a implies that a greater set of nodes is located within the AR for a rebroadcast copy of MSG . From the figure, it can be concluded that a stricter control over redundancy is exercised with increasing values of a , with $a = 0.6$ being the closest to approximating the effectiveness of transmissions based on [8].

¹As before, $a < 1$ since the transmission range is normalized to 1

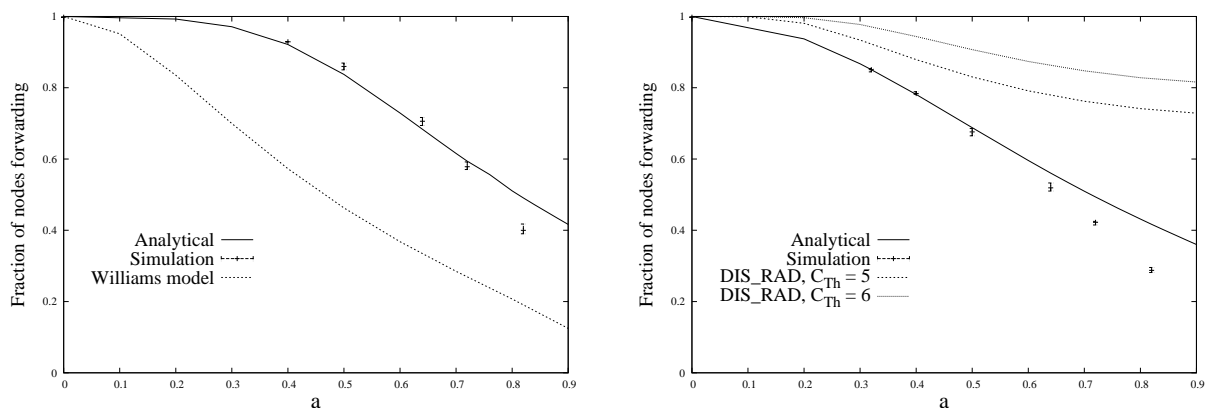


Figure 4.5: Fraction of forwarding nodes when *network information* from (a) only *BRP* is used, (b) both *BRP* and *BOP* is used.

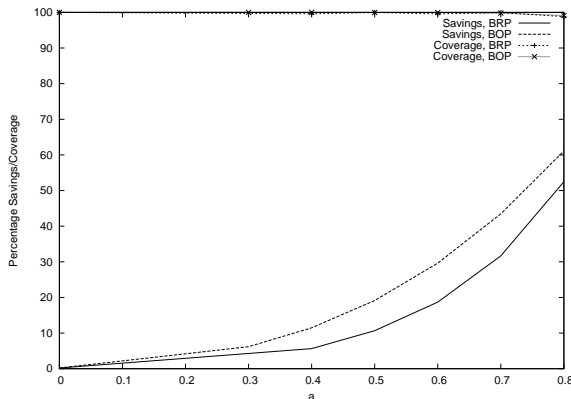


Figure 4.6: Savings achieved at the *broadcast recognition* and *broadcast optimization phases* and the corresponding network coverage.

4.1.3.5 Simulation Results

The author validates the analytical results on the transmission probability of nodes in the context of a broadcasting application. Evaluating broadcasting performance in the context of an application allows the author to examine its impact on other performance metrics as well. The broadcasting application considered here is route discovery for a reactive routing protocol in which nodes decide on whether or not to forward the route request based on *network information* obtained from the *broadcast recognition* and *broadcast optimization phases*. The author uses NS-2 [97] to simulate the routing performance.

The route discovery mechanism is based on the AODV routing protocol [98].

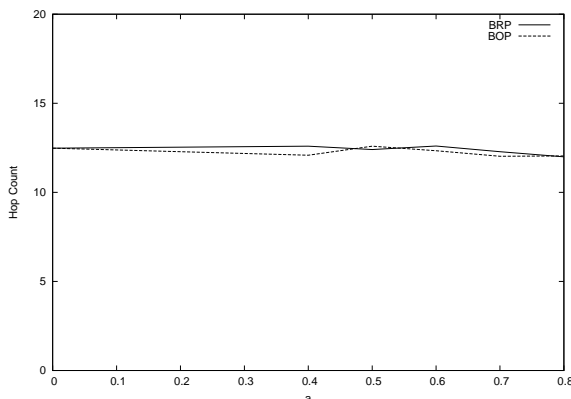


Figure 4.7: Hopcount distribution.

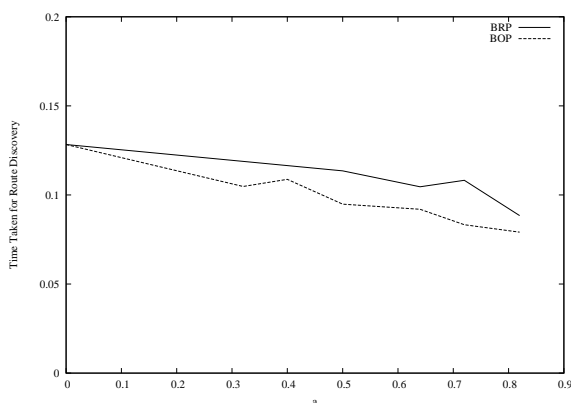


Figure 4.8: Route Discovery Delay

Route discovery is initiated by a source node in the absence of an existing route to the destination by broadcasting a route request (RREQ) after which it waits for the route reply (RREP). In the simulations, the RREQ is propagated throughout the network based on the estimated distance to the packets received in the *BRP* and *BOP*. Results for the different values of a , with $a = 0$ implying pure flooding, are obtained. An IEEE 802.11b network is considered in which nodes contend for channel access using the distributed coordination function (DCF). Results are obtained over 20 topologies each of which a source-destination pair is chosen. The results shown are averaged over 200 initiations of route discovery.

Since the analysis focuses on the performance benefits achievable from *network information* at different stages of broadcasting, it is difficult to compare the proposed

model with existing models that consider individual protocol specific parameters such as the delay [60]. Nevertheless, parallels can still be drawn between the information available during the *BRP* and the analytical model proposed for distance based broadcasting in [60]. For the *BOP*, however, the relation is less straightforward due to the fact that the proposed analytical model considers the entire *network information* available during the *BOP* in terms of the distance and the effect of multiple transmissions. Existing counter based strategies, on the other hand, only consider the number of transmissions overheard. To illustrate the accuracy of the proposed model, therefore, qualified comparison is done by comparing the results for *BRP* with that of the model for distance based scheme proposed in [60].

The simulation results for the fraction of forwarding nodes are shown in Fig. 4.5. The simulation results are shown to match quite closely with the proposed analytical model. The accuracy of the proposed model is particularly highlighted in Fig. 4.5(a) when compared to the model proposed by the authors in [60]. One reason for this is that the analysis does not assume uniform transmission probabilities in the neighbourhood of a node. As the analysis in this chapter is recursive over successive hop-ranges, the transmission probability of a node is obtained as a function of the locations of the nodes in its neighbourhood. Further, the transmission probability is also scaled with the contention among transmitting nodes.

To put in perspective how the performance benefits achievable using *network information* compare with existing algorithms, the performance of the DIS_RAD algorithm proposed in [53] is used. The authors in [53] propose a design which incorporates features of distance based algorithms in counter based broadcasting with the specification of separate random access delays (RAD) depending on whether a node lies within a distance threshold or not. Such a design shares, to an extent, the motivation of utilizing the *network information* available at a node. Note, however, that the performance of the algorithm critically depends on predetermined values of protocol specific parameters of the RAD and the counter threshold (C_{Th}) which do not take into consideration the instantaneous information available at the nodes. Fig. 4.5(b)

compares the analytical performance of DIS_RAD with $C_{Th} = 5, 6$ (ensuring 100% coverage) to the performance achievable with *network information* available during *BOP*, as obtained using the proposed analytical model. The DIS_RAD algorithm results in greater than 70% nodes acting as forwarding nodes. However, a performance of less than 40% can be achieved if instantaneously available *network information* is utilized by nodes, as obtained using the proposed model.

The gap between the analytical and simulation results is seen to be higher for higher values of a . This is due to the approximation factor used in equation (4.6) to account for *Phenomenon-S2*. With an increase in the value of a , the set of forwarding nodes gets more widely spaced out, especially when information from *BOP* is used to determine broadcasting behavior. This implies that the probability that a node in hop range $(h - 1)$ impacts the effectiveness of one in h reduces for higher values of a . The effect is amplified in the case of *BOP* as not only does it serve to reduce the number of forwarding nodes, but it does so by spacing them out further. This results in the impact of $(h - 1)$ on h falling faster than the approximation factor $(1 - a^2)p_t(h - 1)$ used in (4.6). Nevertheless, the accuracy of the results is still quite high when compared to existing studies. Particularly notable is the fact that the analytical models follow the same trend as the simulation results, indicating that they predict broadcasting behavior to a high degree of accuracy.

In addition to the transmission probability, the route discovery performance in terms of the savings obtained is evaluated. As in [8], the savings is obtained as the Savings Rebroadcast Ratio (SRB) which is given as $SRB = \frac{(r-t)}{r}$, where r is the number of nodes receiving the broadcast message while t is the number of them that actually transmit. Fig. 4.6 shows the percentage savings obtained in terms of the number of transmissions as well as the corresponding network coverage. It can be observed that the network coverage does not suffer upon increasing the value of a . Finally, the author observes the effects of reduced number of transmissions on the performance of the routing protocol in terms of hop count. As shown in Fig. 4.7, the hop count does not vary much upon varying the value of a . Thus, use of stateless

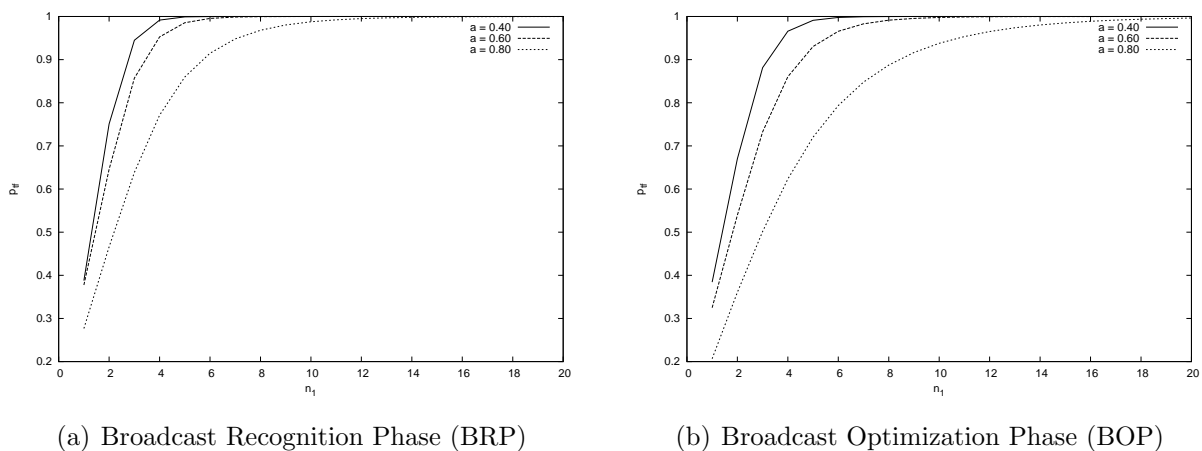


Figure 4.9: Transfer probabilities over varying node densities.

broadcasting algorithms does not degrade the routing performance. Furthermore, the effect of utilizing *network information* reduces the time taken for route discovery, as shown in Fig. 4.8.

4.1.4 Discussion

In this section, the author discusses the implications of the analysis for the design of effective broadcast algorithms. The author uses the results to obtain insights on the feasibility conditions in terms of the broadcast reliability given the network density and the tradeoff between broadcast performance and network costs.

4.1.4.1 Reliability of Broadcasting

The nature of stateless broadcasting implies that it can be used as the benchmark for broadcasting performance in a given network. However, as the performance analysis in the previous section hinges on the choice of parameters for stateless broadcasting, it is necessary to understand how the parameter values depend on the network characteristics. The author evaluates how the node density in a network impacts the choice of the distance threshold a so as to ensure reliability of broadcasting and therefore limits the achievable performance.

To understand the effect of a on reliability, the author focuses on the choice of a

based on the node density so as to prevent the message from dying off. The latter would happen if, at any stage, the transmission of the message from a node is not propagated beyond the immediate one hop neighbourhood. *transfer probability* is defined as the probability that a message transmitted by a node is received by at least one node in its second hop-range. In other words, this is the probability that the message has been propagated beyond the current neighbourhood.

Upon being transmitted by either the source or any other node, the message *MSG* is retransmitted by other nodes in the neighbourhood depending on the transmission probability obtained earlier. The probability that any such transmission from a node located at a distance x is received by a node in the second hop-range can be obtained as $\frac{\pi-A(x,1,1)}{3\pi}$. The probability that any transmission from the first hop-range is received by a node in the second hop range is, therefore, $p_{th} = \int_0^1 \frac{\pi-A(x,1,1)}{3\pi} 2x dx$. Given a node density ρ , the number of nodes in the first hop-range is $n_1 = \pi\rho$ and in the second hop-range $n_2 = 3\pi\rho$. Thus, the probability that a transmission made in the first hop-range is not received by any node in the second one is,

$$p_{2nr} = (1 - p_{th})^{n_2} \quad (4.14)$$

The scenario that the message is not propagated to the second hop-range, thereby resulting in it dying off, can result due to either of two conditions. Firstly, this could result if none of the neighbours in the first hop-range transmits as a result of the choice of a . Denoting the transmission probability as p_{tx} , the probability that none of the first hop neighbours transmits can be obtained as $(1 - p_{tx})^{n_1}$. Subsequently, given that at least one node transmits, the message propagation may still be halted if none of the transmissions is received by any node in the second hop-range. The probability of the occurrence of this event is $\sum_{k=1}^{n_1} p_{2nr}^k \binom{n_1}{k} p_{tx}^k (1 - p_{tx})^{(n_1-k)}$. Thus, the total probability that the message transmitted from a node is not propagated to the

second hop-range is

$$p_{nr} = (1 - p_{tx})^{n_1} + [1 - (1 - p_{tx})^{n_1}] \sum_{k=1}^{n_1} p_{2nr}^k \binom{n_1}{k} p_{tx}^k (1 - p_{tx})^{(n_1-k)} \quad (4.15)$$

The *transfer probability*, is thereafter obtained as,

$$p_{tf} = 1 - p_{nr} \quad (4.16)$$

The value of p_{tf} obtained above expresses the impact of the choice of a on the broadcasting reliability. Thus, the target broadcasting performance for a given node density would correspond to the highest value of a for which $p_{tf} = 1$. The author obtains numerical results for the transmission from the source for increasing node densities. The corresponding *transfer probabilities* for both the *BRP* and *BOP* are shown in Fig. 4.9. Thus, while it is attractive to have an aggressive broadcasting strategy that minimizes the number of transmitting nodes, which corresponds to a high value of a , this is only feasible for very high node densities.

4.1.4.2 Network Costs Versus Broadcast Performance

Since stateless broadcasting operations do not require any prior information at the nodes, an obvious motivation is to investigate the additional benefits achievable if some information were indeed made available. Acquiring such information would require additional transmission overheads, which would need to be offset by the broadcasting performance gains. To illustrate this tradeoff, the author analyzes the effect on stateless broadcasting if the *network information* used earlier were to be itself transmitted explicitly instead of being available as part of the overheard transmissions. An intuitive measure of network costs is the energy consumed by the network and hence the tradeoffs involved in energy consumption are evaluated.

In [99], the author proposes a stateless route discovery mechanism for reactive routing protocols such as AODV. As part of this, the AODV Route Request, AODV_RREQ is accompanied by an additional packet, AODV_RREQ_ALERT, which is transmitted

over a shorter range. All nodes that receive an AODV_RREQ_ALERT along with the AODV_RREQ do not forward the route request while the rest do. Based on this, consider a stateless broadcasting algorithm in which nodes transmit an additional message *ALERT* over a range given by a when forwarding the message *MSG*. A node that receives the *ALERT* in the *broadcast recognition phase* chooses not to take part in the forwarding. A node that receives an *ALERT* during the *broadcast optimization phase* chooses not to forward with a probability p_{disc} . The author examines how transmission of additional *ALERT* messages impacts the energy consumption of the network.

Existing energy models to determine the energy consumption of the network [69] are used. The minimum transmission power required by a node to ensure that the transmitted signal is received correctly at a distance r can be given as $E = t_{rx}r^\alpha$ where α depends on the communication medium and typically takes values between 2 and 4 and t_{rx} is the reception threshold. Normalizing the transmission range to 1 as before, the power consumed at a node due to forwarding a copy of *MSG* would be $E_{MSG} = t_{rx}$ and that of an *ALERT*, $E_{ALERT} = t_{rx}a^\alpha$. In the case of pure flooding which incurs no additional overhead, the energy consumed in the network would be the total energy consumption of all transmissions. The total energy consumed would, therefore, be $E_{Tot} = n_f t_{rx}$ where n_f is the number of transmissions. In the case where *ALERT*s are used, they contribute to the total energy consumption for the network in addition to the forwarded copies of *MSG*. Thus, if $n_a = f_a n_f$ denotes the number of transmissions in the network in such a scenario, the total energy can be given as $E_{aTot} = n_a t_{rx}(1 + a^\alpha)$, where $f_a \leq 1$. Now, for the scenario with *ALERT*s to be feasible, the following condition would need to be satisfied,

$$\begin{aligned} E_{aTot} &< E_{Tot} \\ \Rightarrow n_a t_{rx}(1 + a^\alpha) &< n_f t_{rx} \end{aligned} \tag{4.17}$$

Using $f_a = \frac{n_a}{n_f}$, the author arrives at the feasibility condition for which the mechanism

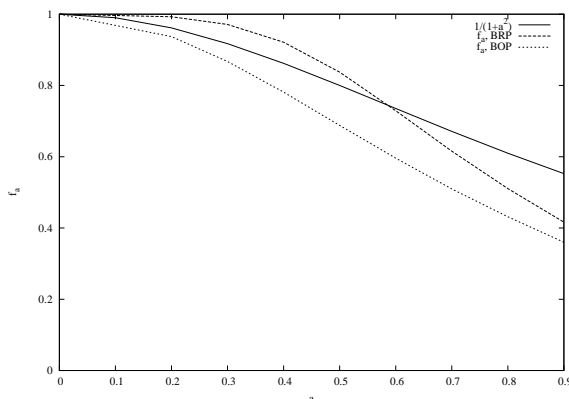


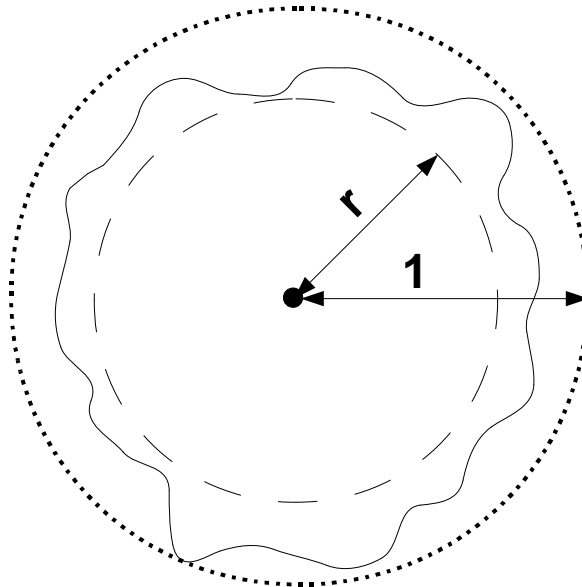
Figure 4.10: Relation of broadcasting performance to feasibility condition.

can be implemented, that is

$$f_a < \frac{1}{1+a^\alpha}. \quad (4.18)$$

This implies that the fraction of nodes that transmit as a result of implementing the proposed broadcast mechanism cannot exceed $\frac{1}{1+a^\alpha}$. Similarly, any stateful broadcasting mechanism can be expected to have a feasibility condition of a similar form if the overheads involved in acquiring network state can be expressed in similar terms as above. As an example, consider stateful protocols where additional neighbourhood information is obtained by exchanging “HELLO” packets. Without going into the specifics of such a protocol, the author can identify the additional costs involved as the number of “HELLO” packets transmitted per node. Thus, in this case, if n_h is the number of “HELLO” transmissions and n_{tx} is the number of broadcasts resulting from the protocol design, the author would have $E_{aTot} = (n_h + n_{tx})t_{rx}$. Plugging its value in the first part of Eqn. (4.17), one can derive the feasibility condition. Alternatively, if one were to map directly to the condition in Eqn. (4.18), the number of “HELLO” messages could be thought of as being transmitted by all the n_{tx} broadcasting nodes. The number of “HELLO” transmissions per broadcasting node would then be $\frac{n_h}{n_{tx}}$ which can be equated to a .

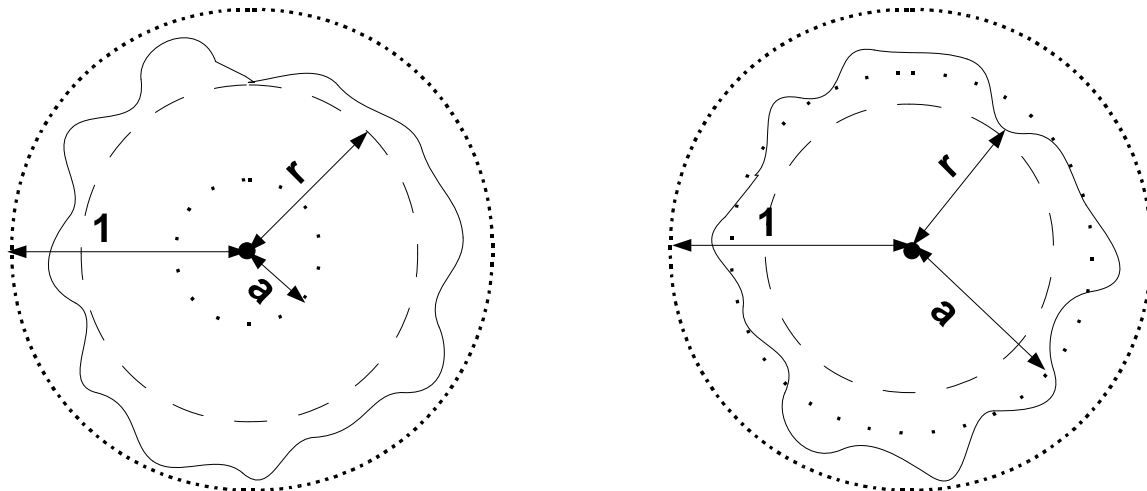
In this case, the author compares the R.H.S. of Eqn. (4.18) to the values of f_a obtained in Fig. 4.10. The value of α is set to 2, which is the most commonly used value. It is observed that, if the information obtained from *ALERT*s is only used

Figure 4.11: Quasi-Unit Disk Model with $0 < r \leq 1$.

in the *broadcast recognition phase (BRP)*, the algorithm is only feasible if $a > 0.6$ although if it is also used in the *broadcast optimization phase (BOP)*, it is feasible for all values of a .

4.1.5 Stateless Broadcasting for Dynamic Channel Conditions

For the analysis so far, the Unit Disk Graph (UDG) is considered, which is popular in wireless networking research due to its simplicity. Such a model allows the author to study the impact of design parameters on the broadcasting performance under idealized channel conditions. However, the assumption of a fixed transmission range is unlikely to be true in actual wireless network deployment due to dynamically varying channel characteristics. Hence, the performance of broadcasting algorithms also needs to be understood in the context of dynamic channel conditions. In this section, the author studies the impact of unstable transmission ranges on broadcasting performance and how the parameter choice can be optimized in such a scenario.

Figure 4.12: Quasi-Unit Disk Model illustrating $a < r$ and $a > r$.

4.1.5.1 System Model

A closer approximation to realistic channel conditions than the UDG model is provided by the Quasi-Unit Disk Graph (QUDG) [100]. Two nodes in a QUDG are connected if the distance between them is less than r , $0 < r \leq 1$ and disconnected if the distance is greater than 1. Nodes may or may not be connected if the distance separating them is between r and 1. The illustration of a Quasi-Unit Disk model is shown in Fig. 4.11. For any pair of such nodes separated by a distance x , $r < x \leq 1$, there exists an edge with probability $\frac{1-x}{1-r}$ [101].

The QUDG model is used to evaluate the effect of the dynamicity of transmission ranges on the broadcasting performance. The author shows analytically how the value of r impacts the reliability of broadcasting for different values of a .

4.1.5.2 Analysis

For a realistic network modeled as a Quasi-Unit Disk Graph, the performance of stateless broadcasting hinges on the values of both the distance threshold a as well as r . To understand this, refer to Fig. 4.12. In the first part of the figure, $r > a$. This implies that in the *Broadcast Recognition Phase (BRP)*, only nodes that are certain to transmit are those located in the region $(a, r]$ while those in the region $(r, 1]$ only

transmit if they receive the original transmission. In the second part of Fig. 4.12, $a > r$, implying that the message MSG is propagated by those nodes located in $(a, 1]$ that receive the message, thereby increasing the uncertainty. The author analyzes the reliability of broadcasting resulting from these two factors. The analysis here is focused on the *BRP* upon which the author later obtains insights for the *Broadcast Optimization Phase (BOP)*.

The transmission probabilities for the $r > a$ case is obtained. In the first hop-range, the probability that any node is located in the region $(a, r]$ is $(r^2 - a^2)$, the transmission probability for which is 1. For any node located at a distance x from the source, $x \in (r, 1]$, the probability of reception of the transmitted message is $\frac{1-x}{1-r}$. Thus, the total probability that any node in the first hop-range broadcasts the message transmitted by the source is,

$$\begin{aligned} p_t^Q(1) &= (r^2 - a^2) + \int_r^1 \frac{1-x}{1-r} 2x dx \\ &= (r^2 - a^2) + \frac{1+r-2r^2}{3} \end{aligned} \quad (4.19)$$

The transmission probabilities for $h > 1$ are obtained in the same manner as before. The same notations as earlier are used. For a node i located in the hop-range h at a distance x from the source, the set of nodes in the hop-range $(h-1)$ from which it can receive the broadcast message is located in the region $A(x, 1, h-1)$. Among the nodes located in this region, all transmissions from the nodes in the region $A(x, r, h-1)$ can be received without error. However, transmissions from nodes located in the region $A(x, 1, h-1) - A(x, r, h-1)$ would be received with a probability $\frac{1+r-2r^2}{3}$. The probability that i refrains from broadcasting upon receiving is that it receives the message from a node located in the region $A(x, a, h-1)$,

$$p_{rf}^Q(h, h-1, x) = p_t^Q(h-1) \frac{A(x, a, h-1)}{A(x, r, h-1) + \frac{1+r-2r^2}{3} [A(x, 1, h-1) - A(x, r, h-1)]} \quad (4.20)$$

Similarly, the total area in the hop-range h from which MSG can be received can be

obtained as $A'(x, h, h) = [A(x, r, h) - A(x, r, h - 1)] + \frac{1+r-2r^2}{3}[(A(x, 1, h) - A(x, 1, h - 1)) - (A(x, r, h) - A(x, r, h - 1))]$. Subsequently, the probability that node i refrains from broadcasting upon receiving a transmission from the hop-range h can be obtained similarly to Eqn. (4.7) as,

$$p_{rf}^Q(h, h, x) = [1 - (1 - a^2)p_t^Q(h - 1)]p_t^Q(h) \frac{A(x, a, h) - A(x, a, h - 1)}{A'(x, h, h)} \quad (4.21)$$

Thus, the total probability that a node in the hop-range h refrains from rebroadcasting the message MSG is,

$$p_{rf}^Q(h) = \int_{h-1}^h p_{rf}^Q(h, h - 1, x) \frac{2x}{2h - 1} + \int_{h-1}^h p_{rf}^Q(h, h, x) \quad (4.22)$$

The probability that a node in hop-range h transmits after having received a copy of MSG can be obtained by substituting $p_t^Q(h) = 1 - p_{rf}^Q(h)$ which is similar to Eqn. (4.9).

For the case $a > r$, it is noticed that in the first hop-range, the probability that a node retransmits the received broadcast is the probability that it is located at a distance x , $a < x < 1$ and also receives the transmitted message. Thus, the transmission probability of a node located in the first hop-range is,

$$p_t^Q(1) = \int_a^1 \frac{1 - x}{1 - r} 2x dx = \frac{(1 - a)(1 + a - 2a^2)}{3(1 - r)} \quad (4.23)$$

For a node in the hop-range h located at a distance x from the source, the probabilities $p_{rf}^Q(h, h - 1, x)$ and $p_{rf}^Q(h, h, x)$ are obtained as,

$$\begin{aligned} p_{rf}^Q(h, h - 1, x) &= p_t^Q(h - 1) \frac{A(x, r, h - 1) + \frac{1+r-2r^2}{3}[A(x, a, h - 1) - A(x, r, h - 1)]}{A(x, r, h - 1) + \frac{1+r-2r^2}{3}[A(x, 1, h - 1) - A(x, r, h - 1)]} \\ p_{rf}^Q(h, h, x) &= [1 - (1 - a^2)p_t^Q(h - 1)]p_t^Q(h) \frac{A''(x, h, h)}{A'(x, h, h)} \end{aligned} \quad (4.24)$$

where $A''(x, h, h) = [A(x, r, h) - A(x, r, h - 1)] + \frac{1+r-2r^2}{3}[(A(x, a, h) - A(x, a, h - 1)) - (A(x, r, h) - A(x, r, h - 1))]$.

Subsequently, the values of $p_{rf}^Q(h)$ and $p_t^Q(h)$ are obtained in the same manner as for $r > a$.

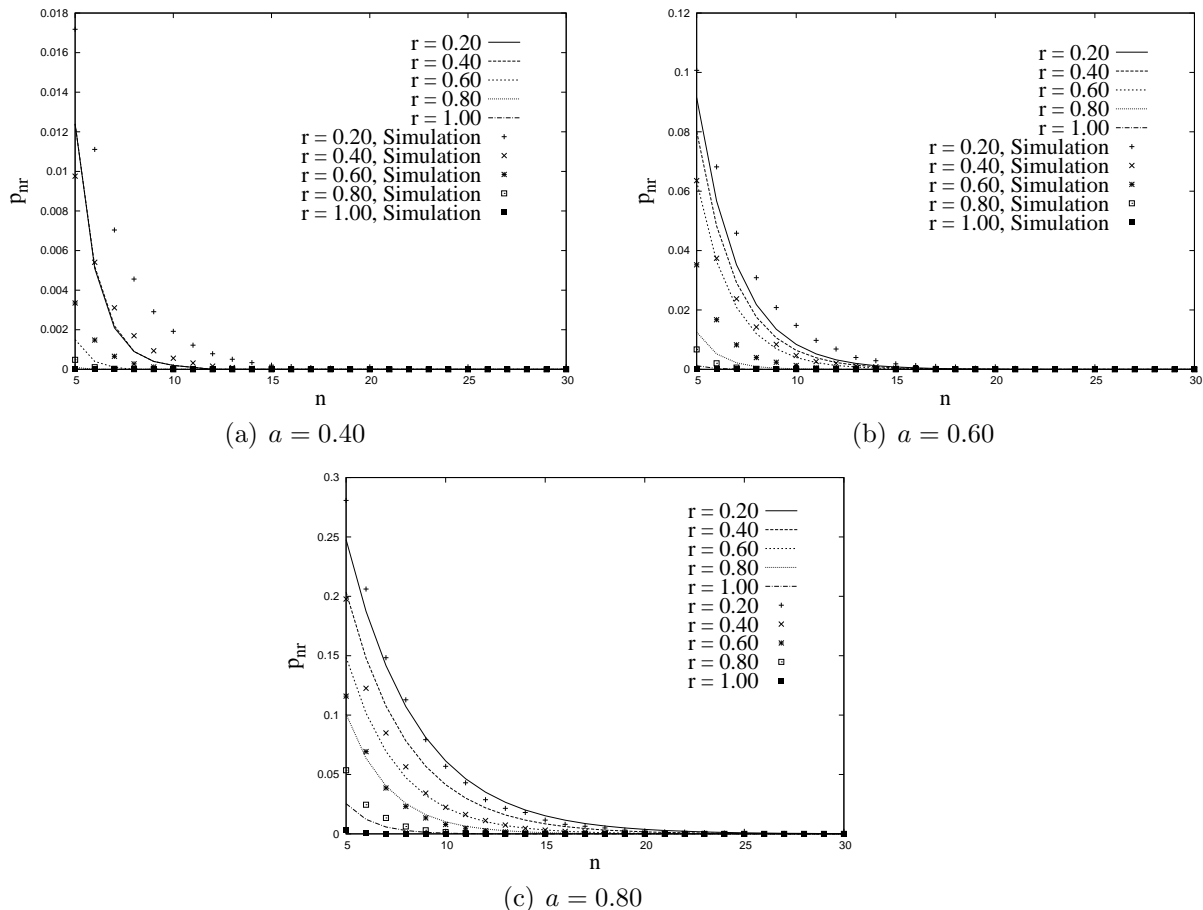


Figure 4.13: Probability that a node does not receive a copy of the broadcast message as a function of a and r .

Having obtained the node transmission probability, the author analyzes how the reliability of broadcasting is affected as a result of variable transmission ranges. The author obtains the probability p_{nr} that a node does not receive a copy of the message MSG even if all nodes located in its first hop-range receive it, i.e. all of them refrain from transmitting. Note that the *transfer probability* defined earlier in Section 4.1.4.1 can also be used as a measure of the reliability of broadcasting in terms of the probability that the message is propagated by the source. p_{nr} is chosen instead to demonstrate the probability of the broadcasting process breaking down irrespective of whether it is propagated by the source.

In the QUD model, the uncertainty of reception of transmissions in the range $(r, 1]$ implies that a node can only receive a fraction of all possible transmissions in its neighbourhood. Thus, given a node density ρ , the total number of nodes in the first hop range is $n = \pi\rho$. Of these, the expected number of transmissions $n(r)$ that a node can receive depends on the value of r ,

$$n(r) = \frac{(1 + r + r^2)}{3}n \quad (4.25)$$

The probability that a node does not receive any copy of the broadcast message MSG is,

$$p_{nr}^Q(a, r) = (1 - p_{tx}^Q(a, r))^{n(r)} \quad (4.26)$$

where $p_{tx}^Q(a, r)$ is the transmission probability of a node.

The effect of the value of r on the reliability of broadcasting is shown in Fig. 4.13 for different values of a . The effect is particularly pronounced for $a = 0.8$ implying that an aggressive broadcasting strategy with a high value of a can only be feasible in near-perfect channel conditions.

4.1.5.3 Insights for Algorithm Design

The results obtained above indicate that the broadcasting behaviour of nodes needs to adapt to the channel conditions. To accomplish such a design, it is necessary for a node to be aware of the channel conditions in its neighbourhood. However, it is not possible for a node to gain a precise estimate of its neighbourhood using the *network information* in the *BRP*. Approaches based on building connected dominating sets (CDS) typically make use of complete local or global topology information. However, in addition to the transmission overheads incurred in the dissemination of neighbourhood information, such a design is not robust to time-varying channel conditions. This necessitates the need to estimate immediate neighbourhood information. An ideal way to estimate network conditions is to make use of *network information* available in the *broadcast optimization phase (BOP)*. As transmissions in the *BOP* are

contextual to the current broadcasting process, instantaneous channel conditions can be gauged more accurately. With this in mind, the author obtains insights on how to optimize node transmission behaviour based on the information available in the *BOP*. The insights obtained could subsequently be used for design of broadcasting protocols which adapt spontaneously to dynamic network conditions.

Let x_{BRP} and x_{BOP} denote the distance over which a node receives transmissions in the *BRP* and *BOP* respectively. In stateless broadcasting algorithms, the effectiveness of a node's transmission is determined primarily as a function of its EAC. As a result, if the transmission from a node i is received by a node in its *BRP* with a small x_{BRP} , then the optimal choice for the latter would be to refrain from retransmission. However, if the channel conditions are bad (i.e. r is low), a low value of x_{BRP} may not imply low EAC as there are likely to be other nodes within i 's range that does not receive its transmission. Thus, it is preferable for nodes not to refrain from forwarding purely on the basis of x_{BRP} . Nevertheless, the effectiveness is still proportional to this value since a transmission received with high x_{BRP} always indicates high EAC.

In the *BOP* itself, transmissions received with high x_{BOP} do not significantly reduce the EAC of the node. However, a high value of x_{BOP} also indicates good channel conditions and vice versa. This implies that for nodes with low x_{BRP} , the discarding probability p_{disc} should be directly proportional to x_{BOP} .

4.1.6 Conclusion

In this section, a comprehensive performance analysis of stateless broadcasting algorithms is provided. The proposed analytical model is based on parameters derived from the information available at the nodes during different stages of broadcasting. As a result of this, the model is generic to all stateless broadcasting schemes. The author identifies how the *network information* at the nodes map to the broadcasting performance benefits. The author uses the proposed model to obtain feasibility conditions for the algorithm to operate in given the network density and costs incurred. Finally, the author shows how the dynamic nature of the wireless channel can be in-

corporated into the existing model. The author also draws insights on how algorithm design can be made robust so as to adapt to varying channel conditions.

4.2 Design of Stateless Algorithms for Multi-Rate Broadcasting

In this section, the author focuses on the design of stateless algorithms for multi-rate broadcasting. The author's contributions are the identification of key design parameters on the basis of different aspects of multi-rate broadcasting. The author clearly establishes how broadcast effectiveness at different rates are related to one another and to the overall broadcast performance. The author uses these insights to develop stateless broadcasting algorithms with the objective of minimizing both the number of forwarding nodes as well as the broadcast latency. The author discusses two algorithms that allow nodes to determine their transmission behaviour in terms of linear and exponential functions of the distance to the broadcasting node and compare them to a counter based mechanism. As the focus of this work is to identify design issues specific to multi-rate broadcasting, the author does not discuss the design of protocol specific parameters such as a random delay [9] typically used to adapt to different network topologies. Nevertheless, the author shows using simulations on NS-2 that the proposed algorithms can provide significant benefits for varying network conditions.

4.2.1 Multi-Rate Broadcasting Analysis

The effectiveness of single-rate broadcasts have earlier been analyzed in terms of the expected additional coverage (EAC) of a node's broadcast [8]. Since any broadcast in the neighbourhood of a node is also received by other neighbours, it results in reduction of the EAC of the node. In a multi-rate scenario, a primary consideration is that data rates have different transmission ranges owing to factors affecting the channel quality required for different rates [11, 102]. The rate-range relationship aggravates

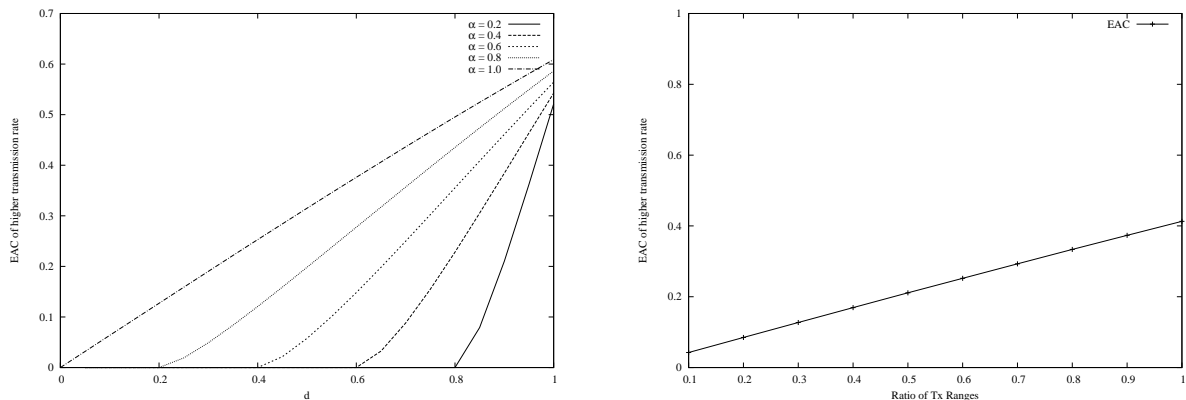


Figure 4.14: Comparison of (a) EAC against distance to a low-rate broadcasting node (d) for different values of the relative transmission range α , (b) Avg. EAC to ratio of Tx. Ranges. $\alpha = 1.0$ corresponds to the lowest rate.

the impact on reduction of EAC. Thus, a broadcast overheard at a low transmission rate results in faster reduction of the effectiveness of transmitting at a higher rate. Here, the author examines the different aspects of how receiving broadcasts at a particular rate could be used to interpret the EAC of a node.

4.2.1.1 Effect of Neighbourhood Broadcasts

Similar to existing analysis of broadcasting in single-rate networks [8, 103], the author analyzes how the transmission effectiveness in terms of EAC is impacted by overheard broadcasts. The author also looks at the impact on the transmission effectiveness of a node at a data rate r_B when it overhears a broadcast at r_A , such that $r_A < r_B$. Thus, $R(r_A) > R(r_B)$, where $R(r_A)$ and $R(r_B)$ are the transmission ranges for r_A and r_B respectively.

Suppose a node B overhears a broadcast at rate r_A from a node A located at a distance k from it. Normalizing $R(r_A) = 1$ and $R(r_B) = \alpha(r_B, r_A) = \frac{R(r_B)}{R(r_A)} < 1$, the

additional area B can cover by transmitting at rate r_B could thus be obtained as,

$$EAC_{r_B}(k) = 2 \left(\int_{\frac{1-\alpha^2(r_B, r_A)-k^2}{2k}}^{\alpha(r_B, r_A)} \sqrt{\alpha^2(r_B, r_A) - x^2} dx - \int_{\frac{1-\alpha^2(r_B, r_A)-k^2}{2k}}^{1-k} \sqrt{1 - (x+k)^2} dx \right). \quad (4.27)$$

The average additional coverage area at rate r_B upon overhearing a broadcast at r_A can be obtained as,

$$E[EAC_{r_B}] = \int_{-1}^{1-\alpha(r_B, r_A)} 2k EAC_{r_B}(k) dk \quad (4.28)$$

The upper limit on the above integral, $1 - \alpha(r_B, r_A)$ is due to the fact that $k > (1 - \alpha(r_B, r_A))$ and would imply that $EAC(k) = 0$ since the entire coverage area of rate r_B would be covered by r_A . The author obtains the above expressions of EAC for different values of $\alpha(r_B, r_A)$. These are shown in Fig. 4.14.

Irrespective of the distance between broadcasting nodes, multiple reception of the same broadcast message would also imply a reduction in the coverage area of nodes. For higher transmission rates with shorter ranges, overhearing multiple broadcasts at a lower rate would lead to greater reduction in the coverage area. The author uses simulation results to understand the effects on reduction in coverage area for different ratios of transmission ranges and show these results in Fig. 4.15.

4.2.1.2 Broadcast Effectiveness Tradeoffs

Owing to the inverse relationship of data rates and their transmission ranges, inherent tradeoffs exist between minimizing the latency and minimizing the number of transmitting nodes. The author discusses these tradeoffs from the perspective of the primary broadcasting objective of maximizing the coverage area.

Consider a network supporting a set of rates $\mathcal{R} = \{r_1, r_2, \dots, r_N\}$ with $r_1 < r_2 < \dots < r_N$. The minimum rate r_1 is denoted by r_{min} in the rest of the discussion. For any node i , the maximum coverage is at the rate r_{min} over the area with radius

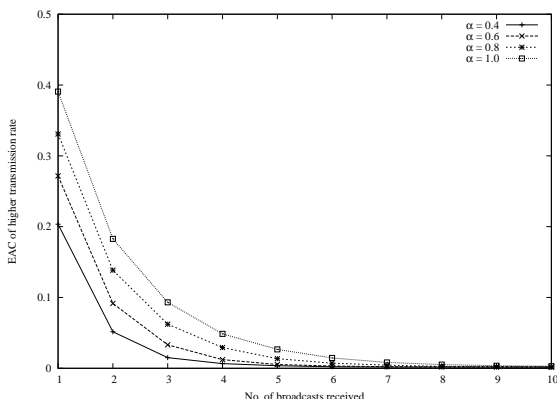


Figure 4.15: Avg. EAC to the no. of received broadcasts for different transmission rates (indicated by the corresponding values of α) relative to the lowest rate.

$R(r_{min})$. From the perspective of optimal broadcasting, the local optima for a node A is obtained by maximizing the part of this region that is covered by the broadcast message. For the minimum latency problem, on the other hand, the local optima is to minimize the time taken to cover the same region. Consider a rate $r_A > r_{min}$ and therefore, $R(r_A) = \alpha_A R(r_{min})$, $\alpha_A < 1$. Thus, if A transmits at r_A , it covers only a fraction of its total neighbourhood, even though the rate of covering the smaller region is higher. As a result, additional transmissions would be required to cover the remaining area. The tradeoff resulting from a high-rate broadcast is, thus, the number of additional transmissions created as a result. Given the author's motivation for designing stateless algorithms, the author defines the number of additional transmissions due to the rate r_A as the number of nodes whose broadcasting behaviour is only a function of A 's broadcast. This is the size of an independent set in A 's neighbourhood, i.e. the set of nodes that can simultaneously access the channel upon receiving the broadcast from A . The choice of an independent set is important here as it refers to a set of nodes that can act as forwarding nodes immediately upon reception of A 's broadcast. Based on this, the number of additional transmissions created at r_{min} , when $\alpha_A \leq 0.5$, is not more than 1. Even though this does not cover the entire neighborhood of A and more transmissions are required to achieve the same, the later transmissions are not counted as created by A 's broadcast since they are likely to be a function of other broadcasts in the network. For $\alpha_A > 0.5$,

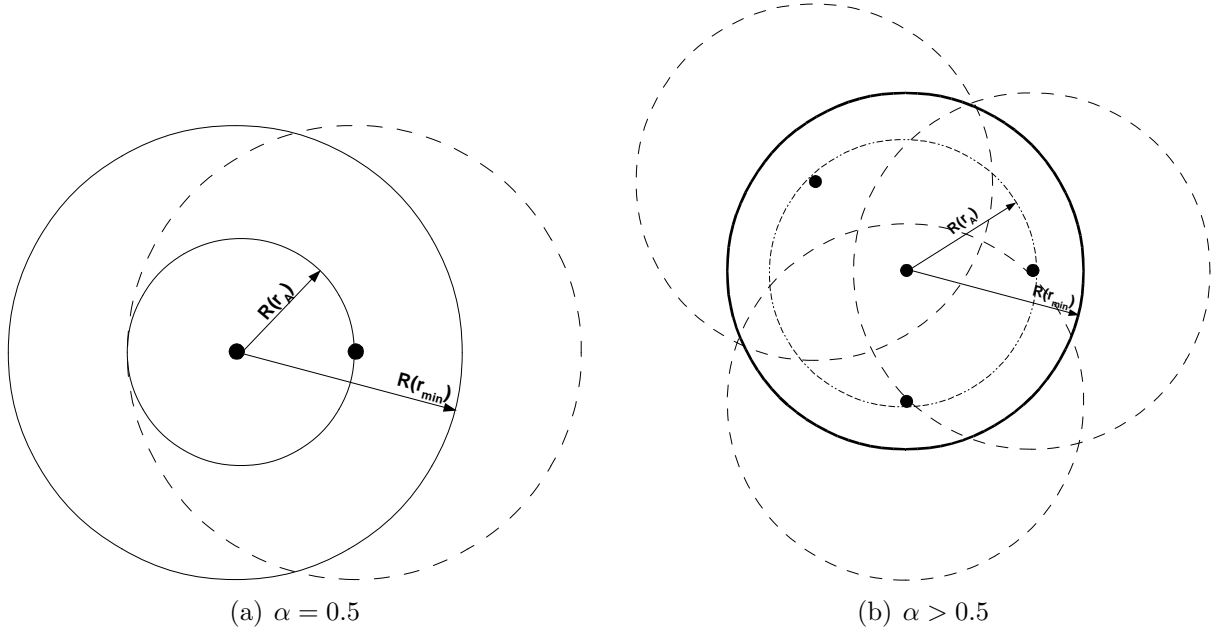


Figure 4.16: Illustration of additional transmissions created at r_{min} .

three transmissions at r_{min} would suffice. Both these scenarios are illustrated in Fig. 4.16. The number of additional transmissions created at rate r_A can be concluded based on earlier results on unit disk graphs [104, 94] as five such nodes. One such construction is shown in Fig. 4.17. Now the ability of these five broadcasts to cover the entire neighbourhood, as in Fig. 4.17, is determined by the value of $R(r_A)$. Here, the author briefly discusses how the conditions for the same can be obtained.

The largest area completely covered between any two such adjacent broadcasts can be obtained as a sector of the circle with radius $\sqrt{3}R(r_A)$. In order to completely cover this circle of radius $\sqrt{3}R(r_A)$, at least one broadcast would be required in addition to those of the five independent nodes. Thus, it can be concluded that the number of additional transmissions created due to r_A is bounded by 6 with the condition $\alpha_A \geq \frac{1}{\sqrt{3}}$. If, on the other hand, $\alpha_A < \frac{1}{\sqrt{3}}$, more transmissions would be required and the problem reduces to the construction of a connected dominating set (CDS), which is NP-complete [96] and has been discussed extensively in existing literature.

Returning to the scenario where $\alpha_A \geq \frac{1}{\sqrt{3}}$, it is observed from Fig. 4.17 that the total area covered actually exceeds the entire neighbourhood of A . This implies that

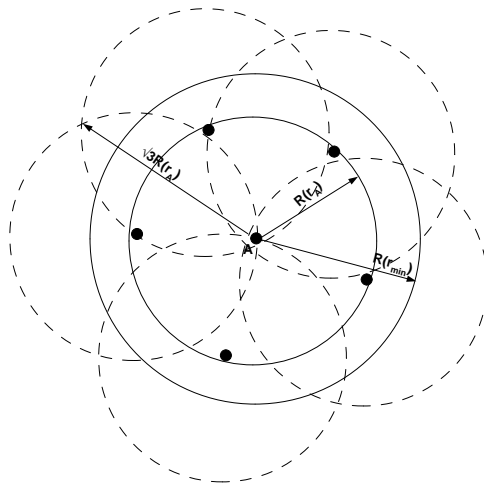


Figure 4.17: Construction showing number of independent transmissions at high transmission rate to cover a node's entire neighbourhood. The dashed lines correspond to the transmission coverage at rate r_A .

the suboptimality in broadcast performance due to multiple high-rate transmissions is offset to a certain extent due to the additional transmissions themselves. The extent of this benefit can be estimated in terms of the additional area covered. From Fig. 4.17, it can be observed that the total area covered would contain most of the circular region with radius $\sqrt{3}R(r_A)$ along with the additional regions containing the petals in Fig. 4.17. Thus, a conservative approximation of the total area would be the circle with range $\sqrt{3}R(r_A)$. Thus, the offset additional coverage (OAC) can be obtained as,

$$\begin{aligned} OAC(r_A) &= \pi(\sqrt{3}R^2(r_A)) - \pi R^2(r_{min}) \\ &= (3\alpha_A^2 - 1)\pi R^2(r_{min}) \end{aligned} \quad (4.29)$$

Since a broadcast from any node apart from the source is the result of an overheard broadcast, the total additional area is in the order of $\Omega(\alpha_A^2)$ times the remaining neighbourhood size.

4.2.2 Multi-Rate Broadcast Algorithms

Three stateless broadcast algorithms are studied in this section. As with earlier algorithms, assume that nodes can estimate the distance to the source of an overheard

Num. Broadcasts Received	Data Rate (Mbps)
1	11.0
2	5.5
3	2.0
4	1.0
≥ 5	Do Not Broadcast

Table 4.1: Choice of Data Rate as a function of the Number of Broadcasts Received transmission based on the received signal strength. Firstly, the author proposes adapting existing counter based algorithms to the multi-rate scenario. Subsequently, the author proposes exploiting the relationship between different data rates and termed this Rate Feedback Broadcasting. Two algorithms, one of which uses a linear feedback function and the other an exponential one are proposed.

4.2.2.1 Counter Based Multi-Rate Broadcasting

The author proposes an adaptation of single-rate counter based broadcasting based on the results of the analysis shown in Fig. 4.15. In the figure, it is observed that effectiveness of higher transmission rates are reduced more drastically than lower ones with an increase in the number of broadcasts received. As the analysis uses the ratio between the transmission ranges of different data rates, an obvious solution is to have the forwarding decision depend on the number of broadcasts received at each data rate. However, this would result in the creation of a very large decision space. Hence, the author proposes a straightforward mechanism that decides on the transmission rate to be used on the basis of the number of broadcasts received, irrespective of their data rates. This decision map is shown in Table 4.1. The broadcast count used as a threshold for each data rate is such that the EAC is at least 5% of the maximum possible coverage. A counter based design, therefore, attempts to maximize the coverage, thereby prioritizing reliability over the performance metrics of forwarding probability and latency. The next set of algorithms is more aggressive as they seek to achieve a better optimization of all the three metrics.

4.2.2.2 Rate Feedback Algorithms

Linear: From the analysis in section 4.2.1.1, it is noted that, once a node overhears a broadcast at a rate r_i , transmission at any rate $r_j > r_i$ would be effective only if the node is located close to the boundary of the region with radius $R(r_i)$. Furthermore, as noted earlier, this would also be proportional to the ratio $\alpha(r_j, r_i)$. The design of the proposed algorithm revolves around these two observations. Here, the author notes that, in realistic network scenarios, the transmission rate-range relationship is unlikely to be static. However, since the proposed design is based on the ratio of transmission ranges between different data rates, which would be affected equally by dynamic channel conditions, the author expects the proposed algorithms to adapt.

Before describing the function, an intuitive condition where transmission at a rate r_j can be discarded altogether is discussed. Having overheard a broadcast from B at a rate r_i , transmission by A at rate r_j would be completely redundant if the region covered by $R(r_i)$ includes the entire region covered by $R(r_j)$ from A. Hence, if $d(r_i)$ is the distance to the source of the broadcast at rate r_i , any rate $r_j > r_i$ can be discarded if $d(r_i) < R(r_i) - R(r_j)$. If $d(r_i) > R(r_i) - R(r_j)$, the effectiveness of A's transmission at r_j would depend on the area not covered by B's broadcast. This, in turn, would be proportional to $e(r_j) = R(r_j) - [R(r_i) - d(r_i)]$.

The proposed algorithm operates by estimating the effectiveness of transmitting at each data rate upon overhearing a broadcast. Since the effectiveness reduces with the number of broadcasts received, a function that behaves in the same manner needs to be constructed. Initially, before any broadcast message is received, the effectiveness for all data rates is set to 1. Subsequently, for the k -th broadcast received at a rate r_i , the effectiveness of a rate r_j is obtained as,

$$T_{eff-lin}^k(r_j) = (1 - \alpha(r_i, r_j)[1 - \delta(r_i)])T_{eff-lin}^{k-1}(r_j) \quad (4.30)$$

where $\delta(r_i) = \frac{d(r_i)}{R(r_i)}$. It can be seen that the first part of the above expression is equal to $\frac{e(r_j)}{R(r_j)}$. Subsequently, the data rate to be used for transmission can be determined either

Data Rate (Mbps)	Transmission Range (m)
11.0	25
5.5	50
2.0	80
1.0	100

Table 4.2: Transmission Range for different data rates

by a fixed threshold value or by a randomized one. The highest rate that satisfies this value is chosen to transmit. In the simulations, the author uses randomized values which allows the author to get a better estimate of the effectiveness of the algorithm.

Exponential: A linear function in terms of the distance is indicative of the expected additional coverage (EAC) for a node. However, a more accurate measure of the effectiveness of a potential broadcast would be in terms of the fraction of the remaining area. Having received a broadcast at rate r_i over a distance $d(r_i)$, this measure would be of the order of $O(1 - \frac{d^2(r_i)}{R^2(r_i)})$. Based on this, the author proposes estimating the effectiveness of a rate r_j as,

$$T_{eff-exp}^k(r_j) = (1 - \alpha^2(r_i, r_j)[1 - \delta^2(r_i)])T_{eff-exp}^{k-1}(r_j) \quad (4.31)$$

for the k -th broadcast received.

In view of the above function, the rate discarding condition also needs to be modified. Restricting $T_{eff-exp}^k(r_j)$ to values between 0 and 1, the rate discarding condition is obtained as,

$$\delta^2(r_i) > 1 - \alpha^2(r_j, r_i) \quad (4.32)$$

which is a stronger condition than in the linear case. For values of $d(r_i)$ for which $EAC(r_j)$ is greater than zero but is still marginal, r_j will not be chosen for broadcast. The rest of the algorithm is the same as in the linear case.

4.2.2.3 Impact of Inaccurate Neighborhood Information

As with existing stateless algorithms, an important aspect of the algorithms proposed here are that they rely on the estimation of distance between nodes. However,

depending on channel conditions, such estimation may not be accurate resulting in nodes estimating distances to be longer than they actually are. As mentioned earlier, such scenarios are expected to be handled using protocol specific parameters such as random delays. Nevertheless, it is still necessary to evaluate how the above mentioned algorithms might get impacted. For the rate feedback algorithms, inaccurate estimation implies that the rate discarding conditions and the transmission effectiveness at different data rates are impacted. Thus, nodes are less likely to discard higher data rates. Further, bad channel conditions are likely to reduce the probability of packet reception, especially at higher data rates. A crucial design aspect of the proposed algorithms is that, in addition to the estimated distance, the transmission effectiveness is a function of the number of broadcasts received. In bad channel conditions, therefore, reduction in the number of received broadcasts results in higher transmission probability. What this implies is that, while bad channel conditions can increase the number of forwards, reliability is preserved as nodes are more likely to transmit. The same analysis holds true for counter based multi-rate broadcasting.

4.2.3 Simulation Results

The author evaluates the performance of the proposed algorithms with simulation results obtained using NS-2. Consider an IEEE 802.11b network. The distributed coordination function (DCF) is used for node scheduling. The transmission rate and range relationship obtained as a result of the TwoRayGround propagation model is given in Table 4.2. An observation here is that the rate-range relationship is different from that used in earlier studies such as [61], which is based on the QualNet simulator. However, the dependence of the author's algorithms on the relationship between different data rates implies that this difference in the absolute values of the transmission ranges would not impact the performance. This feature also implies that the algorithms are robust to dynamic channel conditions. Thus, protocol design in a realistic case can be achieved based on the network interface characteristics as specified by the manufacturer.

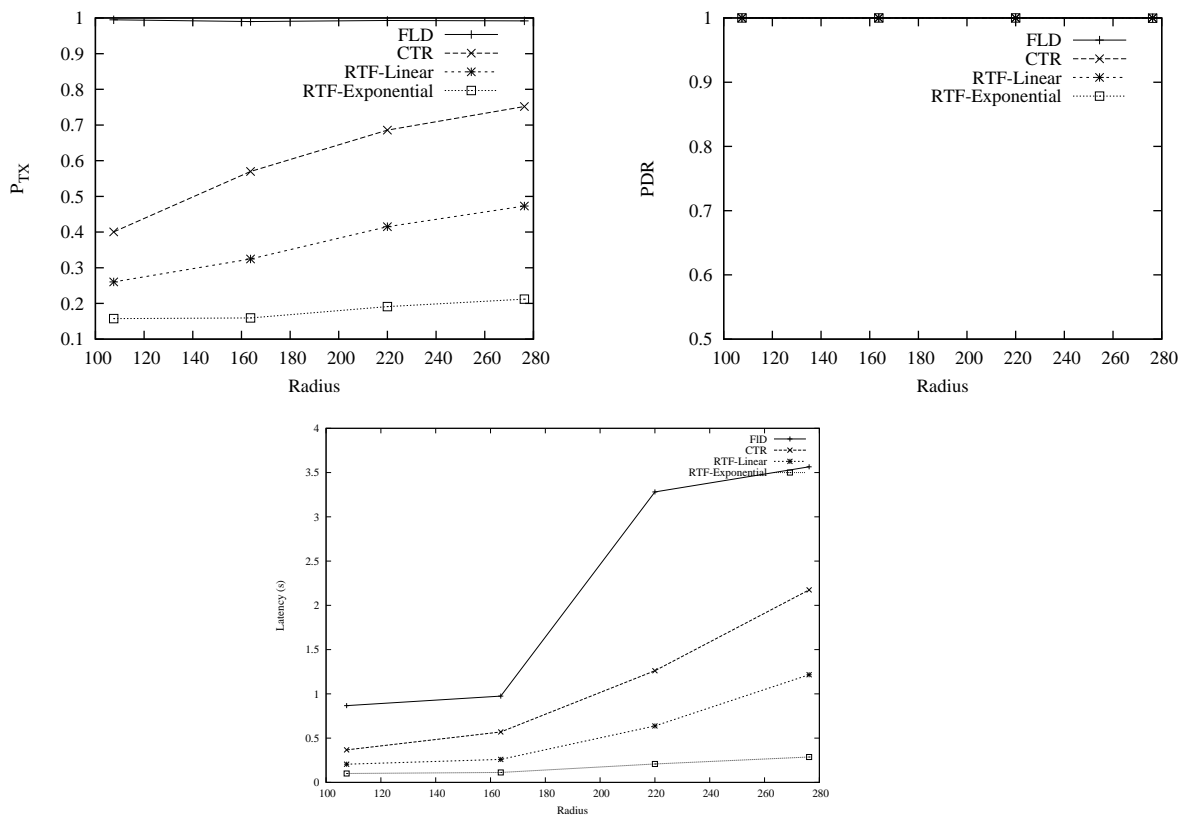


Figure 4.18: Comparison of (a) transmission probability, (b) packet delivery ratio and (c) latency over increasing network size.

The results of the three algorithms are compared with that of blind flooding at the maximum transmission rate of 11 Mbps to allow the author to evaluate the performance of the proposed algorithms relative to the best case performance of flooding. An important point to note here is that the measured latency is the time taken for the entire broadcasting process to complete, and not just the time taken for the message to reach all nodes. This is because, even if all nodes receive at least one copy of the message, it does not give a true indication of the efficiency of the broadcast algorithm since additional copies may continue to get transmitted and thereby, consuming network resources.

The author first studies the broadcasting performance as a function of the geographical size of the network. Consider a setup where the nodes are spread randomly over a circular region, the radius for which is varied to increase the area while main-

taining a constant density. The results for the transmission probability (P_{TX}), packet delivery ratio (PDR) and the broadcast latency are compared with the radius of the simulation region in Fig. 4.18. It is noticed that rate feedback algorithms (RTF-Linear and RTF-Exponential) result in significantly low transmission probability and latency over counter based (CTR) and blind flooding (FLD), while maintaining equivalent packet delivery rates.

In the next set of simulations, the impact of increasing the network density on the broadcasting performance is evaluated. Consider a square region of size 400m x 400m and with varying network density. The corresponding results are shown in Fig. 4.19. It is observed that the transmission probability for the Exponential Rate Feedback reduces with increase in node density. This is due to the fact that the number of broadcasting nodes for a given network is a function of the network area rather than the node density. Thus, a slight increase in the number of transmitting nodes is offset by the increase in the total number of nodes in the network, resulting in a lower transmission probability. In Fig. 4.19(c), it is observed that the latency for the Exponential Rate Feedback mechanism is marginally higher at low densities. This is because, since nodes are likely to be more widely spaced at low densities, the probability of a high rate transmission becomes very small. Correspondingly, there is also a small drop in the packet delivery ratio owing to the fact that randomized threshold values as mentioned in Section 4.2.2.2 are used. The latter implies that, in low densities, the probability that all nodes in the neighbourhood of a node refrain from broadcasting is not negligible. Both the above issues can be addressed in the protocol design by the suitable choice of a forwarding delay [9] and threshold values [8].

Based on the results shown, it can be concluded that the Rate-Feedback mechanisms perform significantly better than counter based and best case flooding for varying network size and density. Moreover, the aggressiveness of the Exponential Rate-Feedback mechanism is ideally suited for moderate to highly dense network scenarios. However, the same results in marginal drop in performance in terms of the

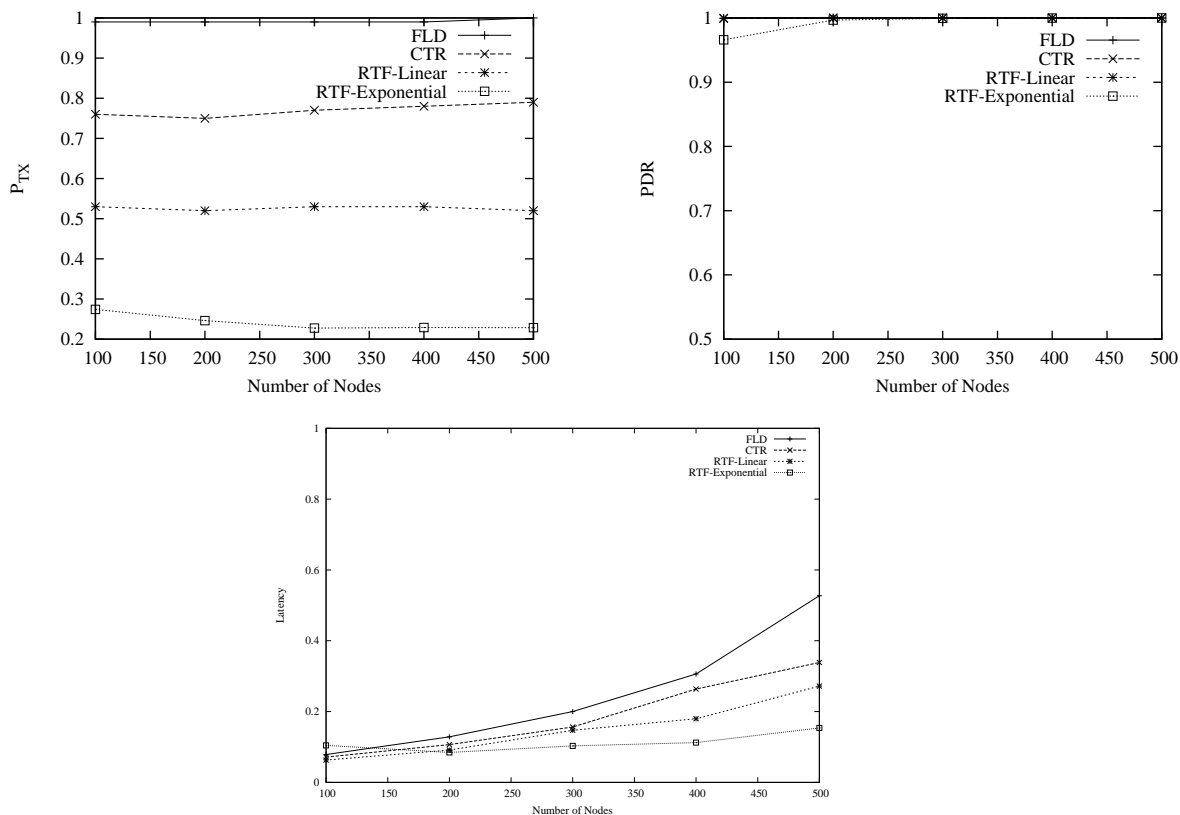


Figure 4.19: Comparison of (a) transmission probability, (b) packet delivery ratio and (c) latency over increasing node density.

latency and packet delivery ratio at low densities which can be handled in protocol design using existing techniques in single rate broadcasting. Contrasted to the Rate-Feedback mechanisms, however, counter based broadcasting is more conservative as it tries to ensure reliability. However, it is still a better choice than blind flooding at the highest data rate since greater than 20% improvement is achieved for both forwarding probability and latency.

4.2.4 Conclusion

In this section, the author studies the problem of network-wide broadcast in a multi-rate scenario. The effect of neighbourhood transmissions on the broadcast effectiveness of a node at different rates is analyzed. Based on these results, the author proposes and evaluates three stateless broadcast algorithms. Simulation results show

that significant performance benefits can be obtained in terms of both minimizing the latency as well as the fraction of forwarding nodes.

The results of this work motivate the author to explore the design of multi-rate broadcasting algorithms in greater detail. The algorithms proposed here can be applied to the design of protocols for multi-rate broadcasting by incorporating earlier findings from stateless broadcasting in single-rate networks. Furthermore, while the current simulation setup considers that nodes are scheduled according to the DCF, the author aims to study node scheduling from the perspective of the Minimum Latency Broadcasting (MLB) problem in a multi-rate scenario.

Chapter 5

A Self-Organization Framework for Wireless Ad Hoc Networks as Small Worlds

The focus of this chapter is to design a framework for wireless networks to self-organize as small worlds using only locally available information. The author studies how small world behavior can be realized in a wireless network by the use of directional beam forming at nodes. The main motivation for using directional antennas primarily stems from the fact that they can be used to transmit over longer transmission ranges than omnidirectional antennas while using the same transmission power. This implies that shortcuts can be created between nodes without the need for additional infrastructure. This distinguishes the author's proposal from existing literature that focus on addition of new links between nodes. In the author's case, however, existing omnidirectional links are rewired as long range directional ones. Another unique property of using directional antennas is that a single beam creates long range links with all nodes that are located within the area covered by the beam. Contrasted to existing designs, this implies that multiple shortcuts are created implicitly, thereby increasing the possibility of reduction in the average path length. Thus, the fraction of long range links in the network is actually higher than the number of nodes beamform-

ing. Furthermore, recent advances in directional antenna design has thrown open the possibilities of large scale deployment in wireless networks, including wireless sensor networks [105, 106]. Therefore, a design considering directional antennas can adapt to a wide range of network scenarios. A potential impact of using directional antennas is that it is likely to result in varying path lengths in different directions for the same pair of nodes, which can affect the performance of real-time data. However, as the focus of the current discussion is to achieve small world properties which results in short average path length in the network, this is expected to guarantee low latency packet delivery.

The author studies the issues surrounding the use of directional beamforming for small world creation in a wireless ad hoc network and proposes ways to achieve an optimal design. Consider a connected network for which the primary objective is to maximize reduction in path length without any loss in connectivity. To the best of the author's knowledge, the only other work where the use of directional antennas has been mentioned for the purpose of creating shortcuts was in [68]. However, like other existing papers, the proposed model considers addition of links using multiple radios.

In order to evaluate the benefits of directional beamforming for realizing small world behavior, a simulation based analysis of the potential benefits and challenges is first performed. A setup in which a fraction of nodes use long range beams in randomly chosen directions. The results show that significant reduction can be achieved in the average path length of the network. However, the path length improvements are accompanied by a high fraction of paths being asymmetric, which results from the directional nature of links [107, 108]. Algorithm design for small world creation using directional beamforming, therefore, needs to strike a balance between path length reduction and loss in connectivity.

Motivated by the benefits achievable using randomized beamforming, the focus is directed to distributed algorithm design for small world creation using directional beamforming. Central to the proposed design is a new measure of centrality that

allows distributed estimation of the structural importance of nodes in the network. Wireless Flow Betweenness (WFB) gives an accurate estimate of the Flow Betweenness Centrality (FBC) [109]. The key aspect of WFB is that it can be computed in a completely distributed manner by exploiting the wireless broadcast advantage (WBA) [69] for information regarding traffic flow, thereby incurring negligible overheads. As far as the author knows, the only other measure of centrality designed from the perspective of distributed implementation in wireless ad hoc networks is the *Aggregated Weight N-hop Ranking (AWeNoR)* proposed by the authors in [110]. However, although it does away with the need for global network information, it still requires explicit multi-hop information.

The author proposes a distributed algorithm that makes use of WFB values computed at individual nodes to decide on their beamforming behavior. Simulations show that this results in greater path length reduction than randomized beamforming with negligible effect on connectivity. Compared to randomized beamforming, the beamforming decision adapts to the structure of the network, resulting in better performance. It is noted here that the discussion does not concern itself with the MAC and routing aspects of directional antennas, which have received substantial research attention in the past. Any of the existing approaches can be used in tandem with the proposed design. More details on the existing approaches can be found in [111, 112, 113] and the references therein.

The rest of the chapter is organized as follows. The next section provides the details of a simulation based analysis of randomized beamforming. Section 5.2 introduces the Wireless Flow Betweenness (WFB). A centralized scheme for choosing optimal nodes for beamforming is shown to give promising results for small world creation. Subsequently, a distributed algorithm for small world creation using beamforming is proposed and evaluated using simulations in Section 5.3. In Section 5.4, a comparative study is done to suitably position the proposed work with respect to existing literature on self-organization and small worlds in wireless networks. Section 6.5 concludes the chapter.

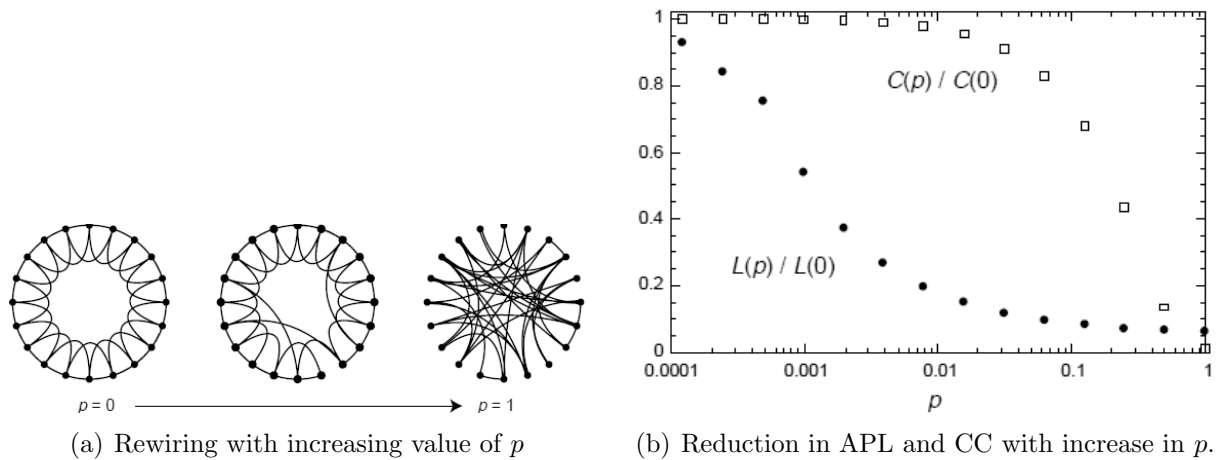


Figure 5.1: Small World properties achieved by random rewiring in a regular network [64].

5.1 Small World Wireless Networks using Directional Antennas

In this section, a simulation based analysis of using directional antennas for small worlds in wireless networks is presented and the results are used to identify crucial design aspects.

5.1.1 Small World Networks

The small world phenomenon was initially studied in the context of social networks by Stanley Milgram [12] who observed that the average separation between a source and target typically lies between five and six, subsequently termed and popularised as "six degrees of separation". It was shown by Watts & Strogatz [64] that by rewiring a small set of links in a regular network, small world behaviour could be realized, characterized by short average path length and high clustering coefficient. Here, the average path length (APL) refers to the mean separation over all possible node pairs in the network. The clustering coefficient (CC) measures the fraction of a node's neighbourhood that are also each other's neighbours [16]. The model proposed in [64] is shown in Fig. 5.1 where all edges in a regular network are rewired with a

probability p . As seen in Fig. 5.1(b), even rewiring only 1% of edges results in a reduction in APL by almost 80% while the CC hardly changes. From the perspective of network design, smaller values of APL imply performance guarantees for data delivery since paths between all node pairs are bounded. This aspect of small world networks was discussed in [114], where it was shown that the diameter of such a network grows with the logarithm of the network size. Kleinberg [115] showed that a decentralized routing algorithm can result in short paths for a grid based model. Further, a decentralized routing protocol for a small world network can be designed with low state information at individual nodes [116]. Coupling of short APL with high values of CC implies high connectivity in the network, thereby making it robust to changes. All these characteristics underscore the attractiveness of small worlds as a self-organization design goal.

5.1.2 Network Model

Consider a wireless ad hoc network of N nodes all of which consist of a single beam-forming antenna. Initially, all nodes transmit using omnidirectional beams with range r . Subsequently, a fraction p of the nodes in the network is randomly chosen which use long range directional beams. Usage of directional antenna by a node can be classified into different categories depending on the modes of transmission and reception [107]. Consider that when a node creates a long range directional beam, it operates in the mode of directional transmission and omnidirectional reception (DTOR). For the purpose of analysis, the sector model for directional antennas [108] is used. The simulation results using the sector model is compared to a more realistic uniform linear array (ULA) antenna [117] model to provide further insight into the research.

A directional beam is characterized by the beam length, width and the beam direction. For the ULA, a longer beam length can be achieved by increasing the number of antenna elements used [117]. By keeping the transmission power constant, increasing the number of elements results in a narrower and longer main lobe, while using a single antenna results in an omnidirectional beam. When using the sector

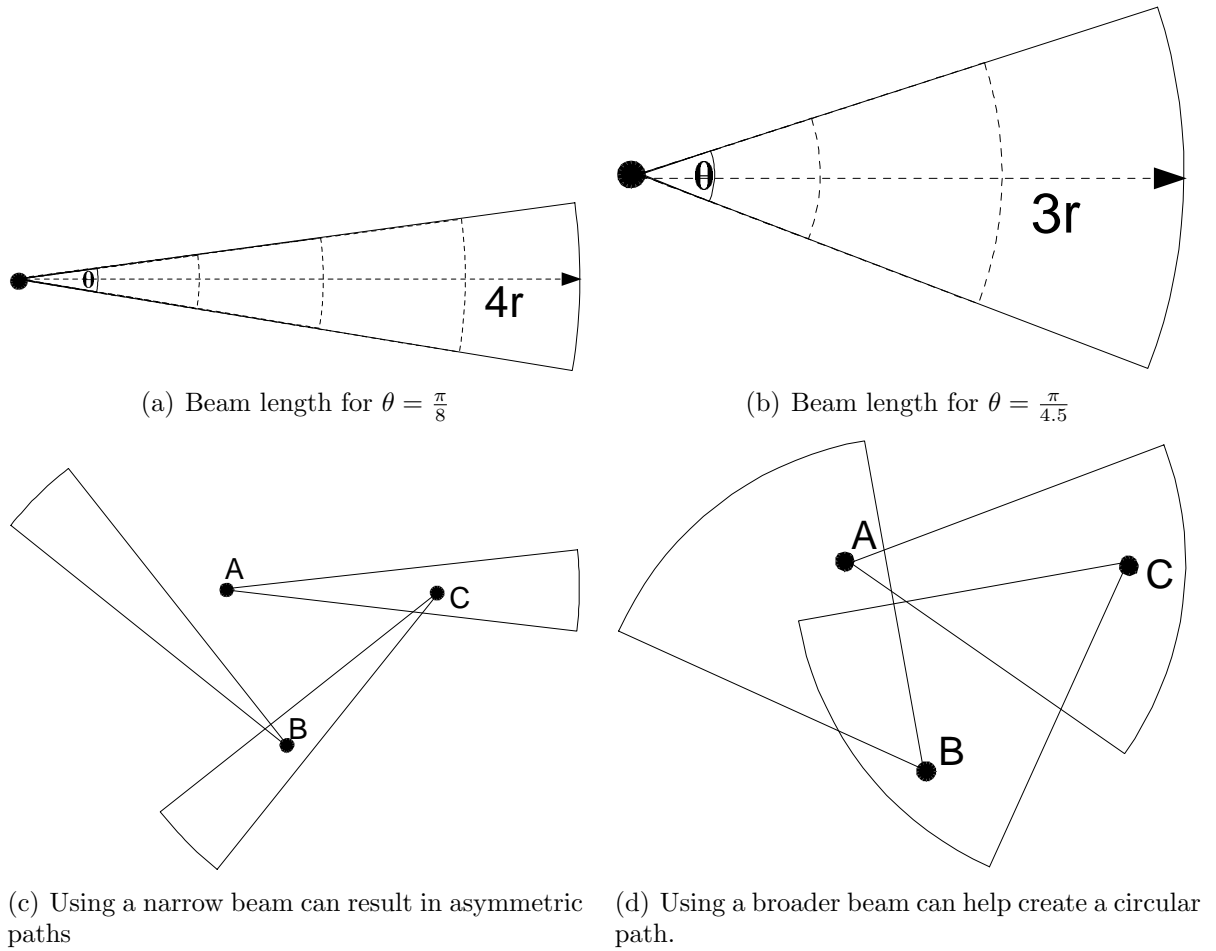


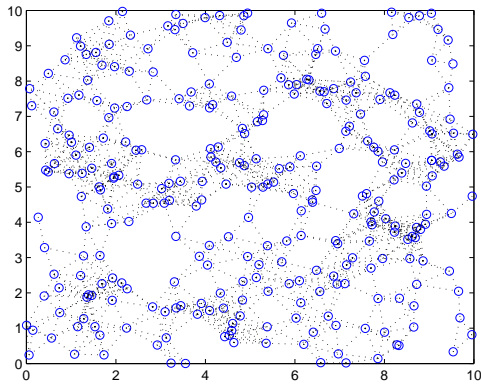
Figure 5.2: Relation between beamwidth and length and the effect on connectivity.

model as an abstraction, a constant transmission power implies that the area covered by the beam stays constant. Thus, the beam length depends directly on the width. The resulting beam length $r(\theta)$ for a beam width θ can be given in terms of the omnidirectional transmission range r as

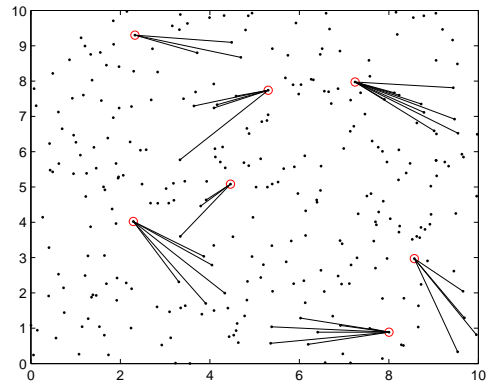
$$r(\theta) = r \sqrt{\frac{2\pi}{\theta}} \quad (5.1)$$

The relation between the beamwidth θ and beamlength is illustrated in Figs. 5.2(a) and 5.2(b).

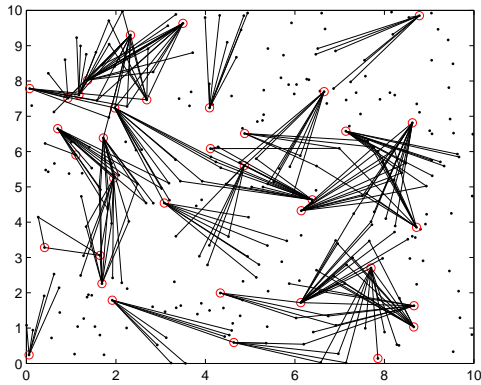
The directed nature of the beam leads can lead to the problem of asymmetric paths between nodes, as discussed in [107, 108]. Ensuring that bidirectional traffic



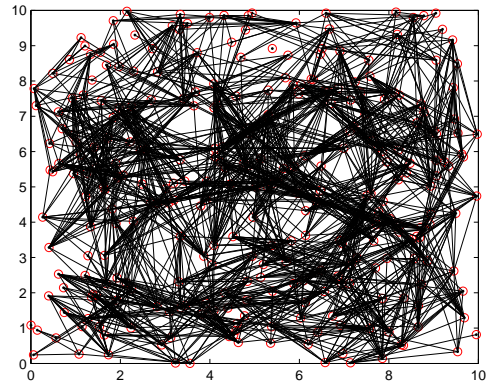
(a) Initial Network Setup with Omnidirectional Beams



(b) Randomized Beams for $p = 0.01$



(c) Randomized Beams for $p = 0.1$



(d) Randomized Beams for $p = 1$

Figure 5.3: Illustration of network with randomized directional beams for different values of p .

flow is supported between a pair of nodes essentially requires the presence of a circular path. Figs. 5.2(c) and 5.2(d) illustrate the effect of beamwidth on the connectivity between two nodes A and B . Using a narrow beam initially implies that transmission can only proceed in a single direction from A to B . Using a wider beam at B ensures that it can communicate with A as well.

To account for the above tradeoffs, a beamwidth that optimizes connectivity and beamlength depending on the density of nodes is chosen. The author optimizes over directional beam lengths that are integer multiples of the omnidirectional range r . To incorporate the tradeoff between increased length and connectivity, the author

divides the sector into separate regions of length r . Subsequently, the beam length $r(\theta)$ is weighed with the probability that at least one node is located in the first and the last regions,

$$r_C = r(\theta)p_{nf}p_{nl} \quad (5.2)$$

where p_{nf} and p_{nl} are the probabilities that at least one node is located in the first and last sectoral regions respectively. The choice of the two probability terms is dictated by the motivation of maintaining connectivity while maximizing beamlength. Since an accurate estimate of maintaining bidirectional connectivity would require knowledge of the entire network, the term p_{nf} is used to estimate the probability that connectivity is maintained with the omnidirectional neighborhood. As the first region under the beam lies within the omnidirectional range, a higher number of nodes here increases the probability that omnidirectional neighbors can be reached. Increasing the beamlength, however, is achieved by reduction in beamwidth. The narrow beamwidth reduces the probability of a node maintaining connectivity to its omnidirectional neighbourhood. The second probability p_{nl} indicates the probability that at least one node benefits from the increased beam length. If there are no nodes present in the last region, greater connectivity can be achieved by increasing the beamwidth while the improvements from the beamlength stay the same. The nodes in the middle regions of the beam are not used in the expression since they do not represent the maximum benefits achievable by increased beamlength nor are they the most affected in terms of connectivity as a result of reduced beam width.

The optimum beam width θ^* among a set of values for θ is chosen as the one that maximizes r_C , i.e.

$$\theta^* = \arg \max_{\theta} \left[r \sqrt{\frac{2\pi}{\theta}} \right] p_{nf}p_{nl} \quad (5.3)$$

The values for p_{nf} and p_{nl} are obtained based on the node density in the network. Given that the number of nodes in the omnidirectional neighbourhood of a node is n , the corresponding values are obtained as $p_{nf} = 1 - (1 - \frac{A_f}{\pi r^2})^n$ and $p_{nl} = 1 - (1 - \frac{A_l}{\pi r^2})^n$ where A_f and A_l are the area of the first and last regions respectively. Recall that

the area under the beam is equal to that of the omnidirectional area when the same transmit power is used.

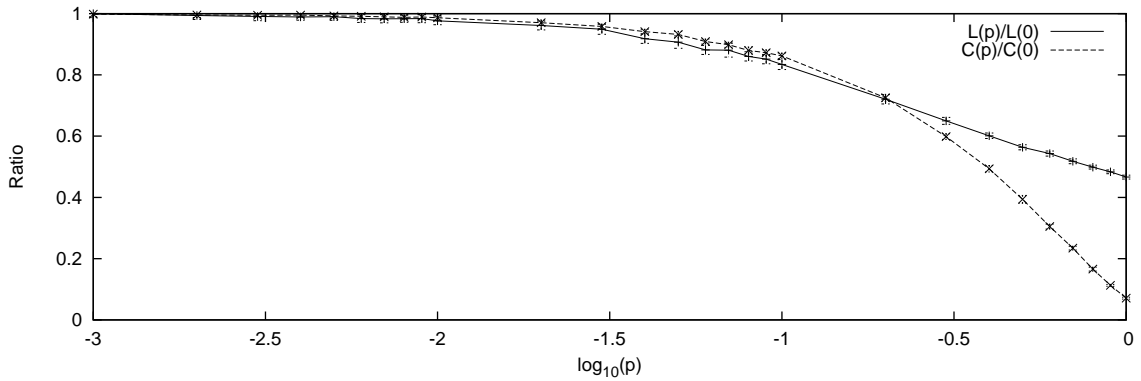
While the sector model is ideal for analysis, simulations are also run on a more realistic model of directional antennas to compare the performance. The model used is that of a uniform linear array (ULA) [117], in which the antenna elements are arranged linearly. The beam pattern of a ULA is characterized by the number of elements used m and the boresight direction θ_b . An important difference with the sector model is that the beam pattern of ULA is characterized by one or more side lobes in addition to the main lobe. The maximum gain obtained in the direction of θ_b is equal to m . In order to map the sector model used earlier to ULA, the number of antenna elements m is equated to $\lceil \frac{r(\theta^*)}{r} \rceil$, r being the omnidirectional transmission range.

Fig. 5.3 illustrates randomized beamforming using the sector model in a network with 300 nodes for different values of p , which is the fraction of nodes beamforming.

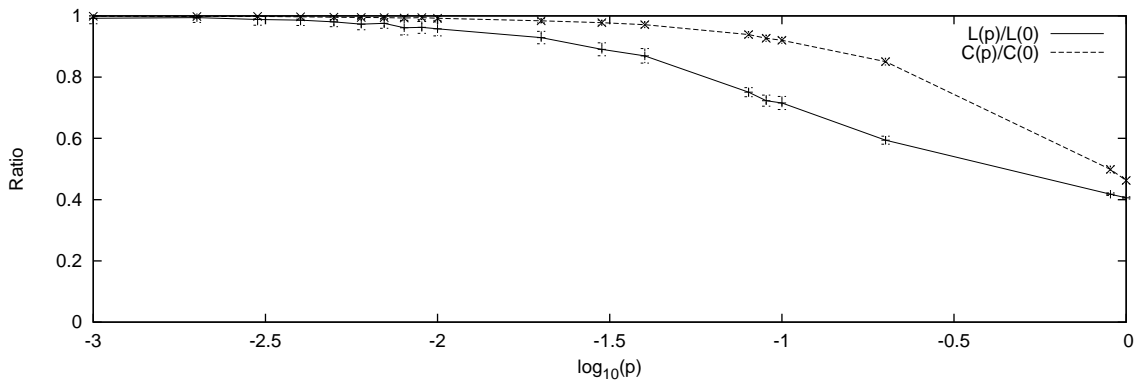
The metrics studied in the simulations are the average path length (APL), the clustering coefficient (CC) and the fraction of node pairs that are connected unidirectionally. The APL is computed by averaging the hop count among all node pairs in the network. Given three nodes u , v and w such that the edges uv and vw exist, the CC identifies the probability that the edge uw also exists. Hence, the CC gives a measure of the cliquishness of the network. Thus, the CC reduces upon rewiring of existing links as it reduces the probability of all edges existing among any three nodes. When using directional beamforming, the rewired edges are also directional, thereby increasing the occurrences of loss in connectivity as a node pair might be connected in one direction but not the other. Therefore, the fraction of node pairs which are only connected unidirectionally is also measured.

5.1.3 Simulation Results

For the simulations, consider a network consisting of nodes using omnidirectional antennas distributed randomly in a rectangular region. The author studies the impact



(a) Sector Model



(b) Uniform Linear Array (ULA) Model

Figure 5.4: Path Length Reduction and Clustering Coefficient as a function of varying the probability of rewiring for $N = 300$

of using randomly oriented directional beams on the average path length (APL) and connectivity of this network. The simulations were run in MATLAB. The results shown were averaged over all possible node pairs in the network for 40 different topologies.

For the first set of simulations, the fraction of nodes p that use directional beams is varied while the rest of the nodes continue to use omnidirectional beams. A value of $p = 0$ corresponds to the initial network setup in which all nodes use omnidirectional beams. The corresponding values of APL and CC are denoted as $L(0)$ and $C(0)$ respectively. The omnidirectional transmission range is normalized to 1 with nodes distributed randomly in a 10×10 region. Fig. 5.4 shows the impact on path length reduction and the clustering coefficient for $N = 300$. The corresponding effect on

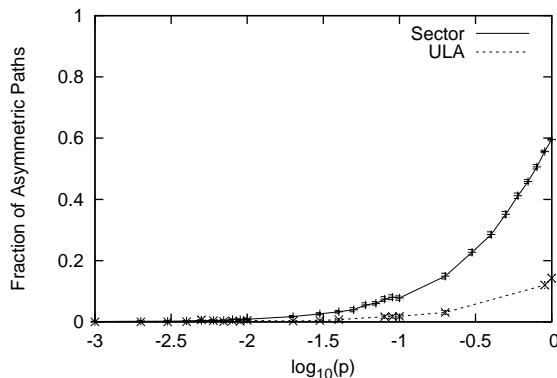


Figure 5.5: Asymmetric Paths as a function of varying probability of rewiring.

asymmetric paths is shown in Fig. 5.5. The results for the sector model, in Fig. 5.4(a), are compared to those using the ULA model, in 5.4(b). The beam length $r(\theta^*)$ obtained using Eqn. (5.3) results in a ratio $\frac{r(\theta^*)}{D} \approx 0.2$. D denotes the maximum distance between any two nodes in the network, i.e. the diameter of the network since r is normalized to 1. In Fig. 5.4(a), the ratio of the reduced path length $L(p)$ and clustering coefficient $C(p)$ to the initial values $L(0)$ and $C(0)$ are shown for the sector model. Note that the values for $\frac{L(p)}{L(0)}$ and $\frac{C(p)}{C(0)}$ are quite close to each other for low values of p . This is contrary to the desired results for small world networks as the path length reduction is accompanied by loss in connectivity. The adverse impact of $C(p)$ on connectivity is seen in Fig. 5.5 as a high percentage of nodes have asymmetric paths. However, when a more realistic ULA model is used, better results are obtained in terms of both the path length improvement and connectivity. The presence of side lobes implies that a beamforming node retains connectivity to a greater fraction of nodes in its omnidirectional neighbourhood, resulting in higher values of the clustering coefficient. This also accounts for shorter path lengths since, in contrast to the sector model, these nodes in the omnidirectional neighbourhood can now be reached in a single hop. As shown in Fig. 5.4(b), the reduction in path length is faster than that of the clustering coefficient. The relatively higher values of clustering coefficient result in better connectivity for lower values of p when using the ULA model as shown in Fig. 5.5. However, as the value of p is increased, the problem of asymmetric paths is

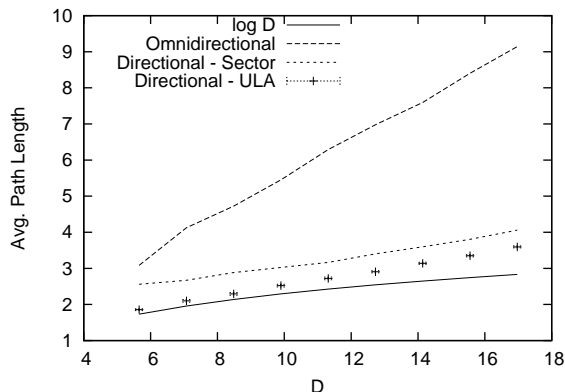


Figure 5.6: Growth of APL with increase in the size of the simulation region.

observed even in this case, though to a lesser extent than with the sector model.

To underscore the suitability of directional beamforming for small world creation, the author views it in the context of existing literature. Helmy [16] obtain results showing the impact of distance limited shortcuts on the path length reduction. In the results shown using the sector model, for the corresponding value of $\frac{r(\theta^*)}{D}$, it is observed that the path length reduction is comparable for small values of p though the reduction is less for higher values. For the more realistic ULA model, though, the path length improvements can actually exceed those of [16].

To observe the relation between the APL to the logarithm of the network size, a set of simulations is run in which the number of nodes is kept constant at $N = 300$ while the size of the simulation region is increased. In Fig. 5.6, the growth of the average path length when all nodes use directional beams is compared to the growth in the value of $\log D$ where D is the largest possible distance between any two nodes in the network. The lines corresponding to the path length for both the sector and ULA models are parallel to that of $\log D$, indicating that the average path length grows as $O(\log D)$.

5.1.4 Discussion and Insights

An important feature of the results in Section 5.1.2 is the tradeoff between path length improvement and connectivity in the network. As described earlier, the choice

of beam length in Eqn. (5.3) is determined by maximizing the probability that a node maintains connectivity with its omnidirectional neighbours. However, as seen in the simulation results, connectivity is still affected severely when a higher fraction of nodes in the network use long range beams.

Based on the results from sector model alone, it is difficult to identify an optimum value of p such that the network exhibits small world behaviour. Using the results for $N = 300$ in Fig. 5.4(a), a 30% reduction in path length can be achieved with $p = 0.2$ but results in about 20% of the node pairs being connected unidirectionally. Ensuring that almost all node pairs are connected bidirectionally would imply a value $p < 0.05$ but the corresponding improvement in path length is less than 10%. However, looking at the results using the realistic ULA model, it is observed that the constraints imposed by the sector model can be relaxed to an extent. The author observes in Fig. 5.4(b) that a 30% of reduction in the average path length can be achieved with $p = 0.1$ while less than 2% of node pairs suffer from unidirectional connectivity.

While the simulations in this section considered randomly choosing nodes to beamform depending on the value of p , the author proceeds to investigate whether greater benefits can be achieved by deterministically choosing nodes to beamform. To do so, the author seeks to identify nodes in the network that are more likely to result in reduction of path length across the network if chosen to beam form.

5.2 A Distributed Definition of Centrality for Wireless Ad Hoc Networks

The structural importance of nodes in a network has traditionally been measured using different notions of centrality [15]. A commonly used measure of centrality is the betweenness [118] which measures the occurrence of a node along paths between other node pairs in the network. High betweenness of a node indicates that it lies along a majority of paths in the network. Creating shortcuts on nodes with high betweenness values is likely to maximize reduction in the average path length in the network as

a majority of paths are affected. Various definitions of betweenness centrality have been proposed in the existing literature. Betweenness centrality, as defined in [118], is known as the Shortest Path Betweenness Centrality (SPBC) as it estimates the importance of a node with respect to the shortest paths between all other nodes in the network. Variants of the SPBC have been reviewed by Brandes in [119]. Everett and Borgatti [120] propose the Ego Betweenness Centrality which is calculated individually by nodes using only their immediate neighborhood information. An alternate class of betweenness centralities that do not consider the shortest paths in the network is proposed by Freeman et al. in [109]. Flow Betweenness Centrality (FBC) measures the importance of a node based on traffic flow in the network. Routing Betweenness Centrality (RBC), proposed by Dolev et al. in [121], generalizes SPBC and FBC by considering paths resulting from routing strategies.

For the problem of distributed self-organization of the network, nodes need to accurately estimate their importance with respect to paths in the network. This constraint makes it unsuitable to use any of the conventional measures as they consider a centralized view of the network. In addition to the distributed computation of the centrality itself, nodes need to decide on their beamforming behavior based on these values. As nodes with high betweenness are ideal for shortcut creation, nodes need to estimate the rank of their betweenness in the network. In order to satisfy these requirements, the algorithm design is centered on a new measure of centrality that allows nodes to decide on shortcut creation depending on the structure of the network.

5.2.1 Wireless Flow Betweenness

The computation of betweenness in a network has been shown to scale as a function of the longest geodesic [122]. In wireless ad hoc networks, the computation costs are particularly costly because of the transmission overheads involved. In order to ensure that network performance is not affected, thus, the computation of a betweenness measure should ideally incur zero overheads. Using this as the motivation, a measure

of betweenness is proposed which is computed by nodes by exploiting the wireless broadcast advantage (WBA), thereby minimizing the overheads incurred.

The broadcast nature of the wireless medium results in implicit sharing of information as nodes can overhear all transmissions in their one-hop neighborhood. Based on this neighborhood information alone, a node can estimate its importance in terms of how often it transmits itself, either as a source or as a forwarding node. A straightforward measure of betweenness of a node v , can therefore be computed as the ratio of the number of times it acts as a forwarding node to the total number of unique traffic flows in its neighborhood. This can be expressed as,

$$w(v) = \frac{g(v)}{\sum_{u \in \{\mathcal{N}(v) \cup v\}} g(u)} \quad (5.4)$$

where $g(u)$ denotes the number of packets forwarded by a node u for distinct source-destination pairs while $\mathcal{N}(v)$ denotes the set of neighbours of v . The denominator gives the total number of packets forwarded in the neighbourhood of v .

The expression in Eqn. (5.4), though straightforward in design, only gives an egocentric measure of importance of the node v . This is because a count of the transmissions in the neighborhood does not give any indication of the likelihood of any flow in the network passing through v . Identifying the exact number of traffic flows in the entire network, including those that do not pass through the neighborhood of v , would be non-trivial and costly. Instead, the author proposes estimating the betweenness of a node by propagating network information as part of traffic flows. The author proposes the Wireless Flow Betweenness (WFB) which uses recursive computation at nodes to obtain an accurate estimate.

The betweenness of a node v based on the set of traffic flows in the network, is defined in [109] as the Flow Betweenness Centrality (FBC) as the share of the maximum flow between all node pairs in the network that passes through v . However, as mentioned earlier, the author cannot adapt FBC directly as individual nodes are not aware of the number of traffic flows in the entire network. A recently proposed

measure of centrality for wireless ad hoc networks [110] requires knowledge of multihop neighborhood of a node, which is again costly in terms of transmission overheads. For the proposed approach, nodes estimate the number of transmissions in the network based on the centrality values of their neighbors. For any node u , the number of transmissions it is aware of can be obtained simply by inverting the expression in Eqn. (5.4) as $\frac{g(u)}{w(u)}$. Thus, a node v can gain an estimate of the number of transmissions in the network using the above estimation for each of its neighbors $u \in \mathcal{N}(v)$. The centrality values of neighbors can be obtained with minimal overheads by having nodes piggyback their centrality values whenever they transmit. However, summing the values for all its neighbors involves multiple redundancies which can result in overestimation.

The first redundancy arises due to the fact that the node v is itself aware of a subset of traffic flows it estimates for a neighbor u with the expression $\frac{g(u)}{w(u)}$. This is because v can overhear all packets transmitted by u , $g(u)$. Thus, node v needs to ensure that the term $g(u)$ is not counted multiple times so as to accurately estimate the number of traffic flows which u is aware of but v is not. Therefore, the number of additional flows that u is aware of as,

$$a_f(u) = \frac{g(u)}{w(u)} - g(u) \quad (5.5)$$

The betweenness of v can be computed using the a_f values of all its neighbors as,

$$w(v) = \frac{g(v)}{o(v) + \sum_{u \in \mathcal{N}(v)} a_f(u)}$$

where $o(v)$ denotes the number of traffic flows overheard by v , including those transmitted by v itself. A node updates its betweenness if it either transmits a packet or overhears one from its neighbors. However, the above summation over all neighbors of v introduces a second redundancy which needs to be taken care of. Just as v is aware of all packets transmitted by a neighbor u , multiple neighbors of v also affect each others betweenness values. Neighbors of v which are also neighbors of each other

estimate the term a_f based on the values of w observed for each other. Consider two nodes u and u' which are neighbors of v as well as neighbors of each other. As part of their centrality calculations, u and u' estimate $a_f(u')$ and $a_f(u)$ respectively. Both these values include the number of transmissions made by v . In the absence of a one-hop neighborhood map, neither node is aware of v being a common neighbor and therefore, $a_f(v)$ is implicitly counted more than once, resulting in overestimation of the number of flows at all the three nodes. This effect is aggravated over multiple iterations and for networks with high node density.

In order to handle the second redundancy, the author limits the set of neighbors whose betweenness values are used for computation. Nodes that infrequently act as forwarding nodes spend a majority of time overhearing transmission from other nodes. Computing the a_f measure for such a flow is likely to result in a lot of redundancy as the majority of the estimated additional flows is likely to be redundant. Thus, the definition of Wireless Flow Betweenness (WFB) is revised to only consider the neighbor which acts as a forwarding node most frequently. The possibility of redundancy is reduced with such a choice as the node chosen is unique within a neighborhood. Furthermore, since a majority of traffic in a given network region flows through this node, it gives a better estimate of the number of flows. Based on this discussion, the WFB value of a node v is obtained as,

$$w(v) = \frac{g(v)}{o(v) + \left(\frac{g(u)}{w(u)} - g(u)\right)} \quad (5.6)$$

$$\text{where } u = \arg \max_{u' \in \mathcal{N}(v)} g(u')$$

A node computes its WFB value whenever it acts as a forwarding node or overhears a transmission in its neighborhood.

By considering a single node in the neighborhood, the chances of overestimation are reduced. However, it also raises the possibility of some flows not being counted. Looking at the denominator of the expression in (5.6), it can be seen, as before, that the term $o(v)$ accounts for all transmissions in the neighborhood of v , and therefore

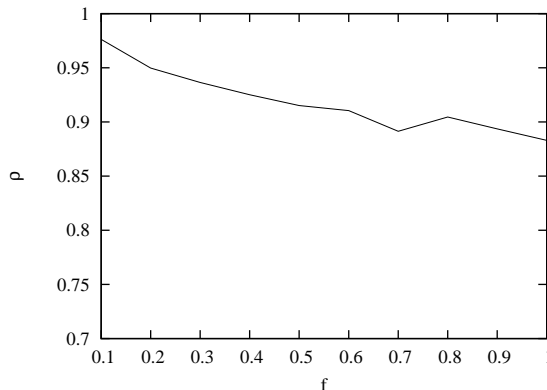


Figure 5.7: Correlation between WFB and FBC for varying traffic load.

includes $g(u)$. Thus, the term $(\frac{g(u)}{w(u)} - g(u))$ gives the number of additional flows in the network estimated recursively. As with node v , the value of $w(u)$ takes into consideration the number of flows overheard by u as well as the node with maximum forwarding count in $\mathcal{N}(u)$. The consideration of the neighbor with maximum forwarding count implies that a majority of flows are always accounted for in a neighborhood, thereby minimizing the probability of flows not being counted. It may be observed that this estimation over successive hops can stop when a node itself has the maximum forwarding count in its neighborhood, leading to an estimation error. On further consideration, however, it can be concluded that such a scenario is unlikely to occur unless the node is the global maxima, i.e. has the maximum centrality value in the entire network. Thus, the possibility of inaccuracies in estimation is minimized.

5.2.2 Correlation with Flow Betweenness Centrality (FBC)

To verify the validity of the proposed WFB measure, the author compares the values obtained with the Flow Betweenness Centrality (FBC) [118]. Since the relative importance of a node with respect to either betweenness measure is given by its rank in the network, the rank correlation of the two measures is obtained. The nodes in the network are separately ranked according to their WFB and FBC values. Subsequently,

the author measures the correlation using the Spearman's correlation coefficient,

$$\rho = 1 - \frac{6 \sum_{i \in V} d_i^2}{n(n^2 - 1)} \quad (5.7)$$

where V is the set of all nodes in the network and d_i is the absolute difference in rank between the two rankings for the i -th node. A value $\rho = 1$ implies perfect correlation between the two rankings.

The amount of traffic in the network is varied and the value of ρ for the WFB is obtained. The FBC values are obtained thereafter. A fraction f of all nodes in the network generate a packet to randomly chosen node as the destination. The results are shown in Fig. 5.7. A very high correlation is seen for lower values of f but it drops slightly for higher values. Since the WFB measure computed does not make use of explicit transmissions to propagate information across the network, as f increases to 1, a greater percentage of information does not propagate over multiple hops. This results in lower values of ρ as f increases. However, even for high values of f , $\rho > 0.88$ implies high correlation between WFB and FBC.

5.2.3 Overhead and Buffer Costs

While the above results show that there is a close correlation between the WFB values computed at individual nodes and the corresponding FBC values, it is necessary to understand the additional costs required for such computation.

The transmission overhead costs are minimal since only WFB values are piggy-backed on to packets by the forwarding nodes, involving one additional field. Thus, additional transmission costs are of constant order. The proposed design, however, requires nodes to store the WFB values of their neighbors along with the corresponding forwarding count. The buffer requirements, thus, scale with increase in the node density. Given a neighborhood size of n nodes, a node needs to store three fields for each neighbor, namely the node identity along with the WFB value and the forwarding count, resulting in a buffer size of the order of $O(3n)$. However, since a lot of

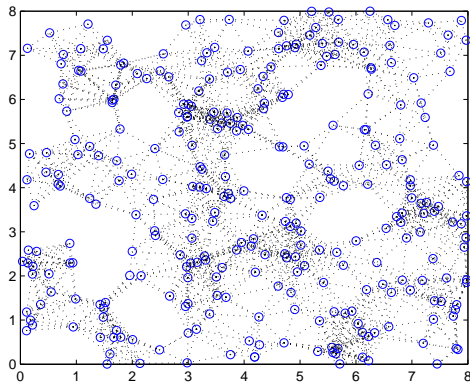
existing ad hoc networks mechanisms rely on presence of neighborhood knowledge, which is of $O(n)$, the additional costs involved for computation of WFB are unlikely to impose a significant burden.

5.2.4 Using WFB for Small World Creation with Beamforming

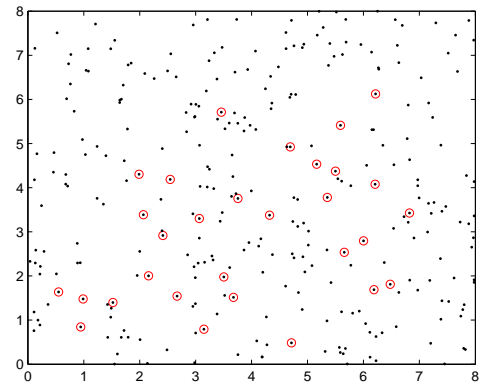
The author now explores the use of WFB for small world creation using directional beamforming. Motivated by the close correlation between WFB and FBC, the performance benefits of using nodes with high values of WFB as beamforming nodes are evaluated. In the current discussion, the author considers that somehow the top ranking nodes with respect to WFB values are identified and they create beams. Based on the insights obtained in this section, the author formulates an algorithm for distributed beamforming in the next section.

Consider that the top 10% (i.e. $p = 0.1$) of all nodes in the network create directional beams. The beamwidth used is the same as for randomized beamforming, formulated in Eqn. (5.3). Nodes orientate their beams in a direction in which they record the maximum hop count based on earlier traffic flows, so as to minimize the network diameter. The simulation results shown here only consider the sector model. Results for nodes distributed at two different node densities are compared. A high node density of $d = 4.6$ nodes per unit area results from distributing 300 nodes over an 8x8 region, and a lower density of $d = 3$ is obtained by increasing the region size to 10x10 for the same number of nodes. Subsequently, the density is kept constant at either of these two values and the size of the region is varied.

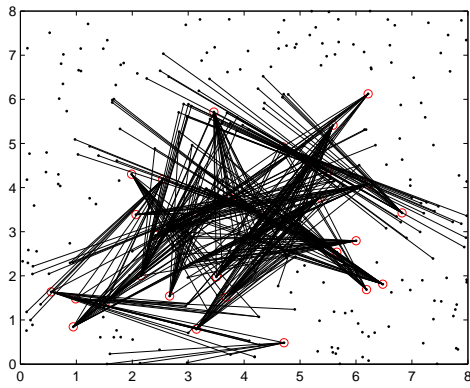
The author shows the performance improvements available for increasing size of the network region, indicated using D as the maximum distance between any two nodes. Fig. 5.8(b) shows the set of nodes chosen as beamforming nodes for the network setup shown in 5.8(a). It can be seen that the majority of the nodes chosen (in red) are located towards the centre of the network. Edges corresponding to the directional beams oriented as specified above are shown in Fig. 5.8(c).



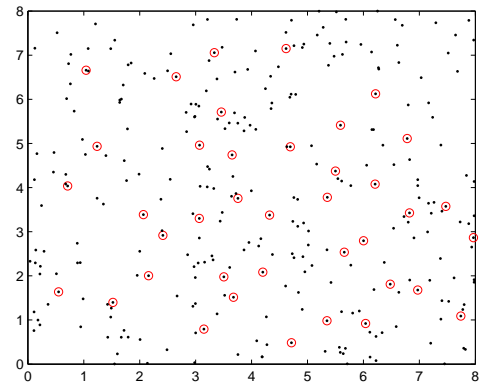
(a) Initial Network Setup with Omnidirectional Beams



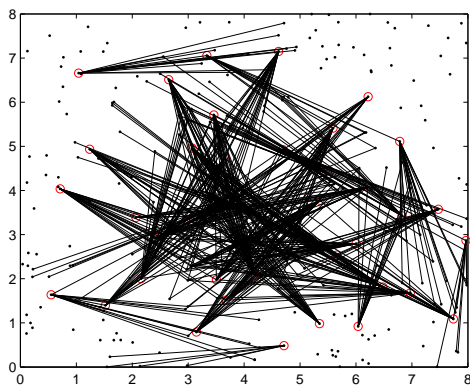
(b) Centralized choice of nodes ranked by WFB values



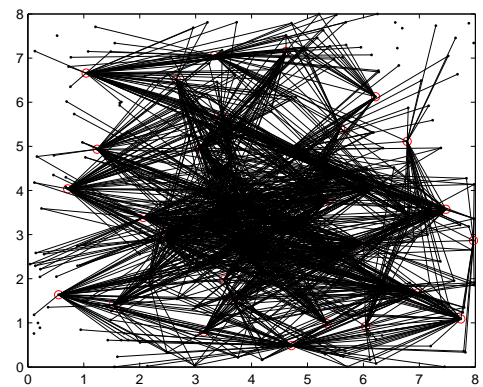
(c) Edges with sector model for centralized choice of nodes



(d) Distributed choice of beamforming nodes



(e) Edges corresponding to sector model for distributed choice of nodes



(f) Edges corresponding to ULA model

Figure 5.8: Illustration of reorganization of network using centralized and distributed choices of beamforming nodes based on WFB.

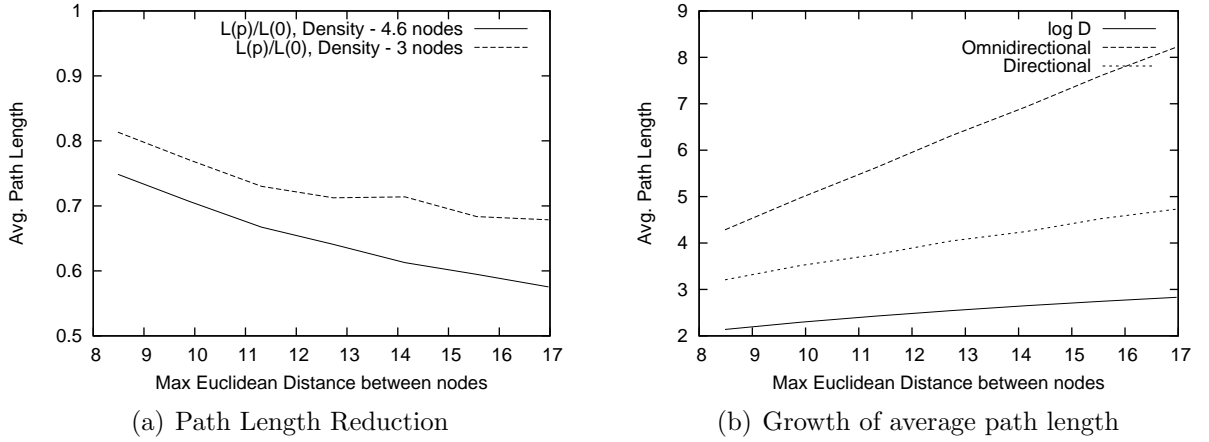


Figure 5.9: Path length reduction by beamforming at nodes with high values of WFB.

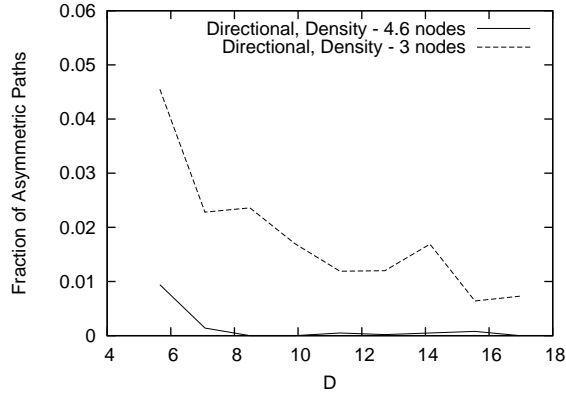


Figure 5.10: Impact on unidirectional connectivity.

Fig. 5.9 illustrates the impact on path length reduction. A reduction of more than 40% and 30% are shown to be achievable for $d = 4.6$ and $d = 3$ respectively in Fig. 5.9(a). Here, it is notable that the improvements in both cases are higher than the corresponding value of p for randomized beamforming, as shown in Section 5.1. As the network size increases, the beam length to diameter ratio reduces to $\frac{r(\theta^*)}{D} \approx 0.2$ for $d = 3$ and $\frac{r(\theta^*)}{D} \approx 0.25$ for $d = 4.6$. Using the results in [16] as a benchmark, it can be seen that the path length reduction is greater for corresponding values of $\frac{r}{D}$. The results in Fig. 5.9(b) compare the growth of the average path length with the logarithm of the network size. The line corresponding to reduced path length using directional beams is shown to grow in parallel with the line corresponding to $\log D$, implying that the growth in path length is logarithmic to that of the network size.

Fig. 5.10 shows the impact of directional beamforming using WFB on the fraction of node pairs that are unidirectionally connected. The effect on unidirectional connectivity is negligible for $d = 4.6$. For $d = 3$, a relatively higher fraction of node pairs are connected unidirectionally, though this is still lower than the value for randomized beamforming. The improvement in connectivity results from the fact that nodes with high value of centrality are better connected. For lower node density, the higher fraction of unidirectional connectivity is due to the choice of beamforming nodes. Nodes with high values of WFB are likely to be neighbors of one another as they are located towards the centre of the network. As the author chooses nodes with high values of WFB as beamforming nodes, they are likely to be neighbors to each other. In the case of lower node densities, this results in a higher fraction of neighboring nodes which are beamforming thereby resulting in unidirectional connectivity. This is alleviated for higher node densities as, in spite of neighboring nodes creating beams, a higher number of neighbors continue to use omnidirectional beams thereby increasing the chances of bidirectional paths.

5.3 Distributed Small World Creation using Wireless Flow Betweenness (WFB)

The author now focuses on distributed algorithm design for small world creation using directional beamforming. The design centers around nodes determining their beamforming behavior based on their estimated importance in the network, thereby adapting small world creation to the network structure. Wireless Flow Betweenness (WFB) as defined above, is used to identify the optimal set of nodes to beamform for achieving small world behavior.

In the previous section, the author has illustrated the benefits achievable by using top ranked nodes by WFB as beamforming nodes. However, as the set of nodes with the highest values of WFB is identified using global network information, it is not suitable for distributed implementation. Furthermore, as discussed before, only

choosing nodes with high values of WFB can result in unidirectional connectivity at low node densities.

5.3.1 Distributed Beamforming Algorithm

The author proposes an algorithm in which nodes decide on their beamforming behavior only based on their own WFB values and those of their neighbors. As nodes broadcast their WFB values as part of packet transmissions, all nodes are aware of the values of their neighbors. As part of the proposed algorithm, a node v decides on using a directional beam if its own value exceeds that of its neighbors by a factor β . Thus, the beamforming condition for a node v can be expressed as,

$$\frac{w(v)}{w_{avg}(\mathcal{N}(v))} > \beta \quad (5.8)$$

where $w_{avg}(\mathcal{N}(v))$ denotes the average WFB for v 's neighborhood.

The above condition ensures that only nodes with high values of WFB choose themselves for beamforming. The fraction of such nodes is determined by the factor β . A higher value of β implies a stricter condition resulting in fewer beamforming nodes. Furthermore, the condition also results in a lower chance of neighboring nodes creating beams. This is because, as a node is chosen if its value exceeds those of its neighbors by the factor β , the condition is unlikely to hold true for any of the neighbors themselves, and therefore excludes them from beamforming. The choice of beam width and beam direction is the same as for centralized choice of beamforming nodes.

The set of nodes that decide on using directional beams is illustrated in Fig. 5.8(d) for the network setup in Fig. 5.8(a). Note that, compared to the centralized choice of nodes in Fig. 5.8(b), the distributed choice of nodes are relatively more spread out, thereby reducing the set of neighbors that simultaneously use beams, resulting in better connectivity. Figs. 5.8(e) and 5.8(f) show the set of edges corresponding to the sector and ULA models respectively. The latter results in a greater set of nodes

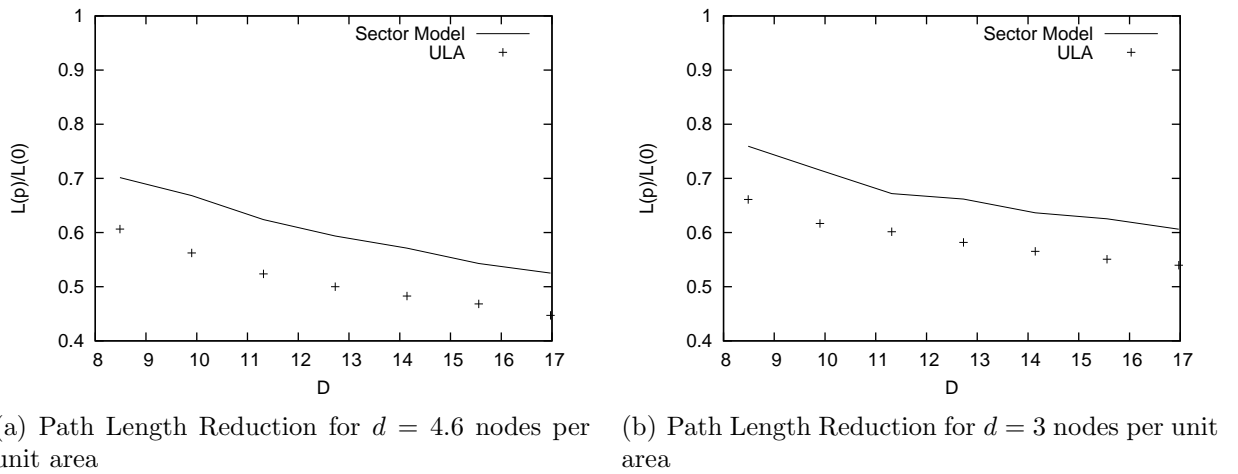


Figure 5.11: Path length reduction by beamforming for distributed choice of nodes.

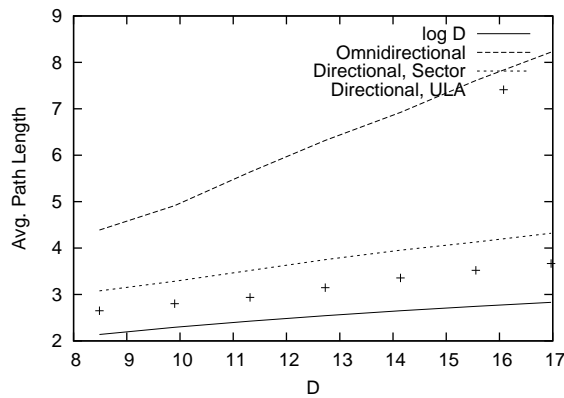


Figure 5.12: Growth of average path length with the logarithm of the network size.

connected using directional beams.

5.3.2 Simulation Results

The proposed algorithm is evaluated using simulations at different node densities as earlier. The value of β is chosen such that the fraction of beamforming nodes stays close to $p = 0.1$, which has been used for the earlier set of simulations using centralized choice of nodes. Using simulations, the author identifies this value as $\beta = 2$ which results in values of p between $0.11 - 0.13$. Results using the sector model as well as using the realistic ULA model are obtained. As in section 5.1.3, the simulation results are obtained using MATLAB. Each individual simulation is averaged over 40

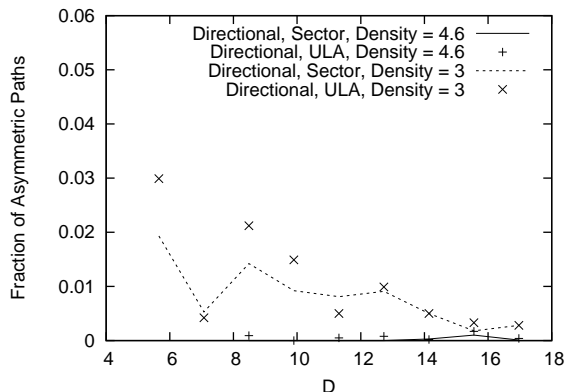


Figure 5.13: Effect on unidirectional connectivity for distributed choice of nodes.

different topologies and the results obtained as the mean over all possible node pairs in the network. A saturated scenario is considered in which every node transmits a packet to a randomly chosen destination.

Fig. 5.11 illustrates the path length reduction for node densities $d = 3$ and $d = 4.6$ nodes per unit area for both the sector and ULA models. When using the sector model, the path length reduction achieved is higher than in the case of centralized choice of nodes. This additional benefit is again due to the fact that fewer fraction of nodes that are omnidirectional neighbors beamform as a result of the proposed algorithm. Directional beams that are located close to each other tend to reduce the improvement in path length as only one of the two may get chosen most of the time, thereby rendering the other redundant. The performance for the ULA model is even more promising as greater than 40% reduction is achieved in both cases, with the improvement being greater than 50% for the greater part of the network region with $d = 4.6$. Fig. 5.12 shows the growth in path length for $d = 4.6$ with respect to the logarithm of the network size. As the choice of beamwidth is the same as earlier, the author obtains similar values of $\frac{r(\theta^*)}{D} \approx 0.2$ for $d = 3$ and $\frac{r(\theta^*)}{D} \approx 0.25$ for $d = 4.6$ when the network size increases. The reduction in path length, is, therefore, much higher than the corresponding values of $\frac{r}{D}$ and p shown in [16].

The impact on connectivity is illustrated in Fig. 5.13. As in the case of centralized choice of nodes, the fraction of node pairs unidirectionally connected is negligible for

higher node density, $d = 4.6$. Likewise, for lower node density where $d = 3$, the impact on connectivity is lower than in the earlier case.

5.4 State of the Art

5.4.1 Self-Organization in Ad Hoc Networks

Self-organization in wireless networks is analyzed and classified in [14, 13]. Dressler [14] classifies existing literature on self-organization mechanisms in ad hoc networks based on the information used for distributed decision making. Prehofer and Bettstetter [13] identify the salient features of a self-organization mechanism. They propose four design paradigms which form the basis for designing self-organization mechanisms for wireless networks. The author discusses how the proposed design for self-organization of wireless networks as small worlds can be associated with the design paradigms in [13] and they are given as follows:

- **Local interactions for achieving global properties:** The author's design centers around the proposed measure of centrality, the wireless flow betweenness (WFB). As the WFB is computed by nodes using only local neighborhood information, this requirement is satisfied. As shown in Section 5.2.2, these local interactions achieve a close correlation to Flow Betweenness Centrality (FBC) computed using global network information.
- **Exploit Implicit Coordination:** Implicit coordination in the proposed algorithm results mainly from the use of the neighbor with highest forwarding count for calculation of WFB in Eqn. (5.6). As such a node is likely to be unique within a neighborhood, all nodes have a common understanding of their WFB values.
- **Minimize long-lived state information:** As nodes update their own WFB values whenever they overhear a packet transmission in the neighborhood, the state information at nodes is fresh with respect to traffic flows in the network.

Parameter/ Algorithm	WFB Based	Guidoni [67]	Sharma [66]	Verma [68]
Shortcut construction	Rewiring	Addition	Addition	Addition
Node Infrastructure	Single radio	High range sensors	Wired	Two radios
Global state information	None	Yes	Yes	Yes
Shortcut edge	Directed	Undirected	Undirected	Undirected
Shortcut direction	Longest recorded path	Random, towards sink	Random	Random
Shortcut length	Function of node density	Constant	Constant	Constant
p	Function of β	Input parameter	Function of network size	Input parameter
Performance metric	Path length, Connectivity	Path length, Clustering coefficient	Path length, Energy dissipation	Path length, Clustering coefficient, Avg. Neighbor degree

Table 5.1: Comparison of existing literature on small worlds in wireless networks

This design can be enhanced further for dynamically changing topologies such as those considering mobility. Thus, instead of storing WFB values of all neighbors, a node needs only store values for those which have recently acted as forwarding nodes. The current discussion is restricted to static networks and hence does not address this aspect.

- **Design protocols that adapt to changes:** In the proposed design, both the choice of nodes as well as the beam direction is determined by traffic flows in the network. Thus, it can easily adapt to an increase or decrease in the number of flows in the network.

5.4.2 Small Worlds in Wireless Networks

Small world creation in wireless networks has been first investigated by Helmy in [16]. Simulation results are used to study the behaviour of wireless networks as a result of random addition of distance limited shortcuts. It is noted in [16] that, owing to the fact that wireless networks are spatial graphs rather than relational, shortcut links

cannot be completely random as in [64]. Rather, the possibility of creating a short cut link between two nodes is determined by the distance between them and the radio transmission range.

The author compares the main features of the proposed algorithm to existing literature on creating small worlds in wireless networks (identified by surname of the first author and corresponding reference) in Table 5.1. Note that the author uses unidirectional connectivity as a performance metric instead of the clustering coefficient as the former gives a more precise measure of the network performance with regard to the use of directional antennas. Another crucial metric is that of energy dissipation, considered in [66]. In the context of the discussion in this chapter, this is not included as a performance metric since the transmission power for nodes using omnidirectional and directional beams is constant.

5.5 Conclusion

In this chapter, the author explores the use of directional beamforming for self-organization of a dense wireless ad hoc network as a small world. A simulation based analysis of the achievable performance benefits of randomized beamforming is presented and the challenges involved are identified. Subsequently, a distributed algorithm for nodes to decide on their beamforming behavior is presented. The author also defines a new measure of betweenness centrality, Wireless Flow Betweenness (WFB), which is used to identify beamforming nodes. Simulation results show that significant performance benefits can be achieved over randomized beamforming.

As part of the future work, the author would like to investigate the navigability of the small world network achieved using the proposed design. As with existing studies on small world networks, decentralized routing algorithms can be designed to identify short paths in the network [115]. As the small world creation algorithm proposed here makes use of WFB values at nodes, routing can also be similarly designed. As with existing literature in SWNs, such a design needs to minimize the amount of global information available at nodes. Furthermore, the author would like to study the use

of the WFB measure for other wireless network protocols that can benefit from the knowledge of the centrality of nodes.

Chapter 6

Exploiting Wireless Broadcast Advantage as a Network-wide Cache

In this chapter, the author is interested in obtaining a measure of the wireless broadcast advantage in a multihop scenario. As discussed above, WBA is a function of the information available implicitly at nodes. For a multihop wireless network, the author obtains a measure of WBA in terms of the amount of information implicitly propagated across the network. To quantify the information available at nodes, the author considers that nodes store all the information received from the overheard transmissions. The storage of all information propagated across the network results in the creation of network wide distributed cache. This implicitly created storage is termed as *broadcast cache*. The author studies the growth of the *broadcast cache* in a network to obtain a measure of the WBA.

Since WBA is characterized purely as a function of the implicit propagation of information, it is necessary to define a suitable framework for such a study. This is done so in the context of exploiting WBA for packet flows. Traditionally, packet delivery over multiple hops is done by explicitly routing the packet through suitably chosen nodes that relay the packet. However, as relaying constitutes an altruistic activity on

the part of the relaying node, it does not involve exploiting the WBA. The author therefore, considers implicit propagation of information as the potential spread of information from non-altruistic transmissions in the network. From a selfish perspective, a node would only be interested in transmitting the data packet generated by itself. The minimum set of non-altruistic transmissions in the network, thus, consists of the set of source transmissions. However, if each of these transmissions were to be combined by each source with all the information received from its neighbourhood, it would result in an implicit propagation of the information.

For the setup, the author considers that each node in the network has a source symbol to be transmitted to some other node. Whenever a node is scheduled for transmission, it transmits a random combination of all the data packets previously received from its neighbourhood with the symbol generated by itself. The focus is not on the specific coding scheme which performs this random combination. Rather, the proposed analysis is focused on the propagation of a symbol due to non-altruistic transmissions in the network that exploit the broadcast advantage. The scheduling of a node thereby determines how far its source symbol can be propagated as a result of the remaining transmissions in the network. As nodes store all overheard packets, the contents of the *broadcast cache* at a node determine its ability to decode specific source symbols. In other words, they determine the number of additional transmissions that would be required in the network for the node to successfully decode a source symbol.

The primary contribution of this chapter is to obtain a lower bound on the number of packets stored in the *broadcast cache* of a node that contains a particular source symbol x_i . Subsequently, the conditions determining whether opportunistic encoding of multiple flows is beneficial for a source symbol is obtained. The proposed analysis is based on an ideal network scenario that considers that node scheduling is done uniformly and transmissions are error free. For a given coding scheme, a receiver is able to correctly decode a particular source symbol whenever it obtains the required number of encoded packets. This is consistent with existing studies such as those on linear network coding which consider that source packets can be decoded once a full

rank of vectors is obtained.

6.1 Wireless Broadcast Advantage

In order to understand the potential benefits of WBA, the author first formulates a relationship between opportunistically received information and its effect on the throughput performance of network flows. Maximizing the throughput of a network flow is equivalent to identifying the optimal source-destination path, as characterized by a path metric such as the expected transmission count (ETX) [123] or the expected transmission time (ETT) [124]. Generalizing the metrics for a single rate network, the author can formulate the performance objective of any network flow x_i , with source $s(x_i)$ and destination $d(x_i)$, as,

$$\max \frac{L(x_i)}{T(x_i)}$$

where $L(x_i)$ is the payload size and $T(x_i)$ is the total information transmitted in the network pertaining to flow x_i . The term $T(x_i)$ is defined as the set of information transmitted in the entire network as determined by the protocol specifications for successful delivery of packets belonging to flow x_i . The exact information constituting $T(x_i)$, therefore, is contextual to protocol specifications. For example, for traditional ad hoc routing protocols such as DSR and AODV, this includes the overheads involved in route discovery as well as route maintenance in addition to the actual packet payload [4]. In geographic routing, on the other hand, information transmitted for location updates constitute significant overheads [5].

Next, the author introduces the concept of *minimum information* that needs to be transmitted for a flow x_i . The minimum information is defined as the set of information that must compulsorily be transmitted irrespective of the protocol used. As discussed above, for successful packet delivery between $s(x_i)$ and $d(x_i)$ using reactive routing, the initial source transmission from $s(x_i)$ needs to be accompanied by fixed and varying overhead costs consisting of protocol specific data, forwarding by intermediate nodes and multiple retransmissions due to packet losses. Alternatively,

techniques such as network coding can be used to combine x_i with other flows in the network [73], in which case, some overheads such as the total amount of forwarding by intermediate nodes may be reduced. Thus, while the total information $T(x_i)$ depends on the protocol used, the initial transmission from the source $s(x_i)$ is common to all scenarios. Therefore, the minimum information, in this case, is the transmission from the source node received by its immediate neighbors. The distinction between actual payload information and protocol information was the focus of [125], where the author proposed design of routing protocols for small transactions in wireless networks by eliminating the need for network wide flooding. However, the proposed algorithm still involved additional information to be propagated across the network. This study analyzes how the minimum information resulting from all non-altruistic transmissions constitute the WBA. As discussed above, this is limited to the set of transmissions from the source nodes. WBA is analyzed as the information propagated implicitly across the network as part of these transmissions.

For the flow x_i , let \mathcal{P}^i be the set of disjoint paths available from the source $s(x_i)$ to the destination $d(x_i)$. Thus, if all the available paths are utilized, the total information transmitted would be a sum of the information transmitted on each path, $T(x_i) = \sum_{P \in \mathcal{P}^i} f_P(x_i)$. Now, let $G(\mathcal{P}^i)$ be the minimum information transmitted over all paths $P \in \mathcal{P}^i$ resulting from all flows in the network. The additional overheads resulting from flow x_i over \mathcal{P}^i can, therefore, be given as,

$$O(x_i) = T(x_i) - G(\mathcal{P}^i)$$

To see how $G(\mathcal{P}^i)$ relates to WBA, note that transmissions from additional flows over the paths $P \in \mathcal{P}^i$ could potentially be used to satisfy flow requirements of x_i since they share common paths [73, 76]. To understand this, suppose that nodes along all paths $P \in \mathcal{P}$ store the information $G(\mathcal{P}^i)$ as received from overheard transmissions. The additional transmission overheads required to satisfy the flow requirements of x_i would now be a function of this opportunistically received information. This is denoted as $O(G(\mathcal{P}^i))$. The total information transmitted in the network pertaining

to x_i could now be given as,

$$T'(x_i) = G(\mathcal{P}^i) + O(G(\mathcal{P}^i))$$

To understand the potential benefits of WBA, consider that a node $v \in P, P \in \mathcal{P}^i$ stores a fraction $\alpha_v(x_i)$ of x_i as part of the opportunistically received information $G(P)$. When the flows using the same paths are combined, the destination $d(x_i)$ requires a minimum required set of information $J(x_i)$ in order to successfully receive the source symbol. For example, when linear network coding is used, the destination needs to acquire a full rank of all encoded vectors to successfully decode any of the original symbol. Now, in the case of traditional routing, the flow x_i would typically be routed along one of the paths $P \in \mathcal{P}^i$. Thus, the information transmitted in the network would be,

$$f_P(x_i) = \sum_{h=1}^{H(P)} t(h)L(x_i)$$

where $H(P)$ is the number of hops from $s(x_i)$ to $d(x_i)$ over the path P and $t(h)$ is the total number of transmissions in hop h . However, since the information $G(\mathcal{P}^i)$ opportunistically received stored by intermediate nodes can potentially be used for successful reception of x_i , $d(x_i)$ can retrieve the necessary information from a set of nodes V located nearer to it such that $\sum_{v \in V} \alpha_v(x_i) = J(x_i)$, rather than have it routed through an entire path P . The difference in the number of transmissions required with that of traditional routing, therefore, constitutes the benefits achievable from WBA.

Thus, utilizing the opportunistically received information G can result in improved throughput due to fewer transmissions. The focus is on analyzing the opportunistically received information as the result of the minimum set of non-altruistic transmissions in the network. Allowing all nodes to store and propagate overheard information results in a network-wide cache called *broadcast cache*.

Existing research has illustrated the benefits of using cooperative caching for improving data access performance in wireless ad hoc networks [126]. Subsequent re-

search has focused on optimal caching strategies [127] and identifying the optimal set of nodes for caching [128]. The author's definition of *broadcast cache* corresponds to a similar notion of cache as it can similarly be used to optimize data access performance. However, unlike existing studies, *broadcast cache* is created implicitly by exploiting transmissions in the network using WBA. In the next section, a network model which results in the creation of the *broadcast cache* is presented and subsequently its growth analyzed in terms of the propagation of information in the network. Section 6.3 discusses the conditions which determine that the implicitly created *broadcast cache* can result in improved data access performance for a flow when compared to traditional routing.

6.2 Broadcast Cache

6.2.1 System Model

Consider a network of N nodes in which each node i generates an independent source symbol x_i . As described earlier, the minimum set of transmissions consists of exactly one transmission from each source, and therefore consists of N transmissions. Owing to the broadcast nature, every transmission is received and stored by all neighbours. Upon gaining access to the channel, a node transmits a random combination of its own symbol with the set of received packets. The author distinguishes between the terms *source symbol* and a *packet* as thus: a *source symbol* refers to the native symbol generated by a source while a *packet* denotes the random combination of such symbols transmitted by a node in a time slot t . Each transmission is assumed to take up a single slot. Thus, if a node i gains access to the channel in the time slot t , the transmitted packet can be given as $y_i = g(x_i, \mathcal{E}_i(t-1))$ where $\mathcal{E}_i(t-1)$ is the set of all packets received till the previous time slot and $g(\cdot)$ is an encoding function that combines x_i with all received packets $z_i \in \mathcal{E}_i(t-1)$. It is assumed that all packets transmitted in the network are of the same length. As a result of this, even though each node transmits only once, the corresponding source symbol can be propagated

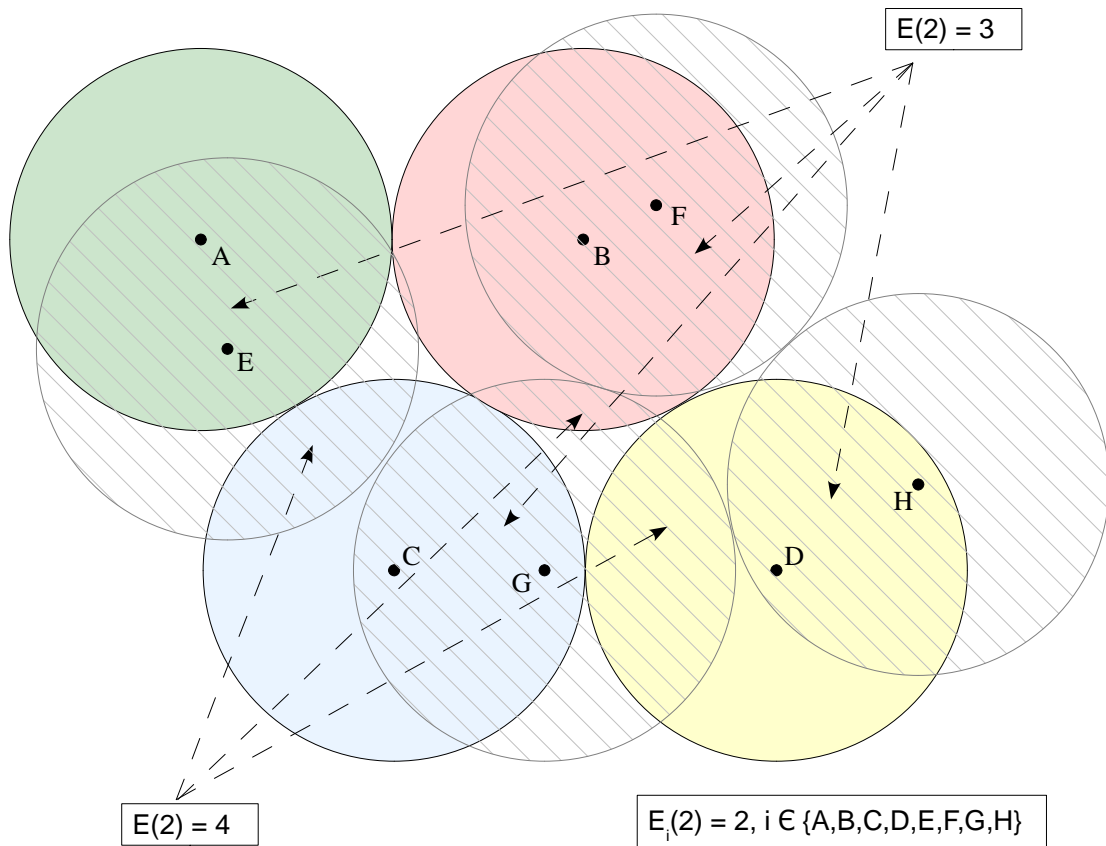


Figure 6.1: Illustration of the growth of *broadcast cache*. $E(t)$ indicates the number of symbols stored in the broadcast cache of a node after t time slots. Nodes with transmission coverage marked by solid colours transmit at $t = 0$ while those with shaded coverage areas transmit at $t = 1$. This results in growth in the number of *constituent symbols* of neighboring nodes. The number of *constituent symbols* stored by nodes in each region is indicated by the dashed arrows pointing at them. All nodes are assumed to generate a source symbol each, which is counted as a *constituent symbol*.

over multiple hops. Fig. 6.1 illustrates creation of broadcast cache as a result of transmissions over two successive time slots. The number of symbols encoded in the set of all packets received by a node i is denoted by E_i , which is called *cache length*. The set of symbols encoded within the packets stored in a node's cache, including its own source symbol, is termed the *constituent symbols* of the cache.

The author focuses on the creation of broadcast cache from collision free transmissions in the network. To satisfy this requirement, it is assumed that node scheduling is done according to optimal broadcast scheduling and any two nodes scheduled for

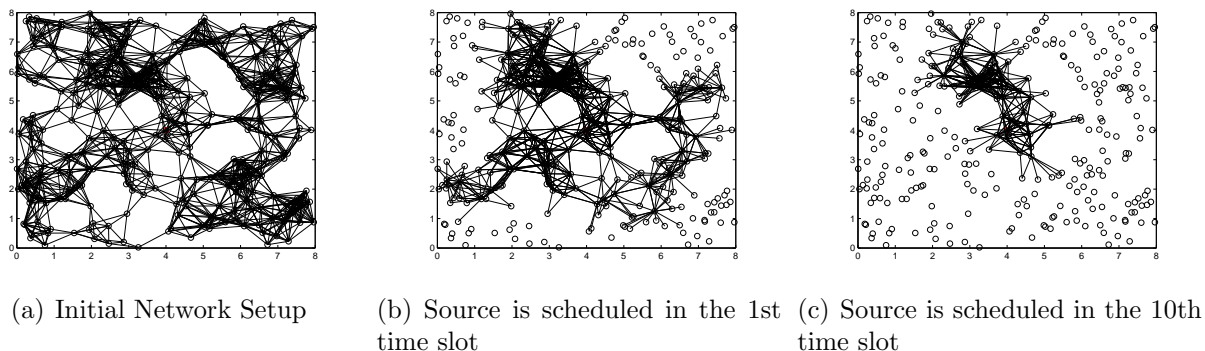


Figure 6.2: Propagation of a symbol with the source located at the center of the network.

transmission in the same time slot belong to a 2-independent set (2-IS) [95] in a unit disk graph (UDG) [96] realization of the network. This ensures that, in any particular time slot, a node is within the range of not more than a single transmission. Transmissions are assumed to be collision free.

6.2.2 Analysis of the Distribution of Broadcast Cache

The authors obtains a lower bound on the probability that a source symbol x_i transmitted by the node i is propagated over h hops after the minimum set of transmissions take place by considering only the propagation of information between successive hops.

The analysis studies the impact of scheduling of a source node on the probability that the corresponding symbol gets propagated across the network. Fig 6.2 shows the propagation of a symbol as a function of the scheduling of the source node. Fig. 6.2(a) shows the initial network setup. Considering the source symbol generated by a node located in the center of the network, Figs. 6.2(b) and 6.2(c) show the propagation of the symbol when the source node is scheduled in time slots 1 and 10 respectively. The nodes linked by dark edges are those that receive the source symbol once all nodes in the network get scheduled. The scheduling of the source is seen to significantly impact the propagation of the symbol. When the source is scheduled in the tenth time slot ($k = 10$), the number of nodes that receive the symbol is a fraction of that

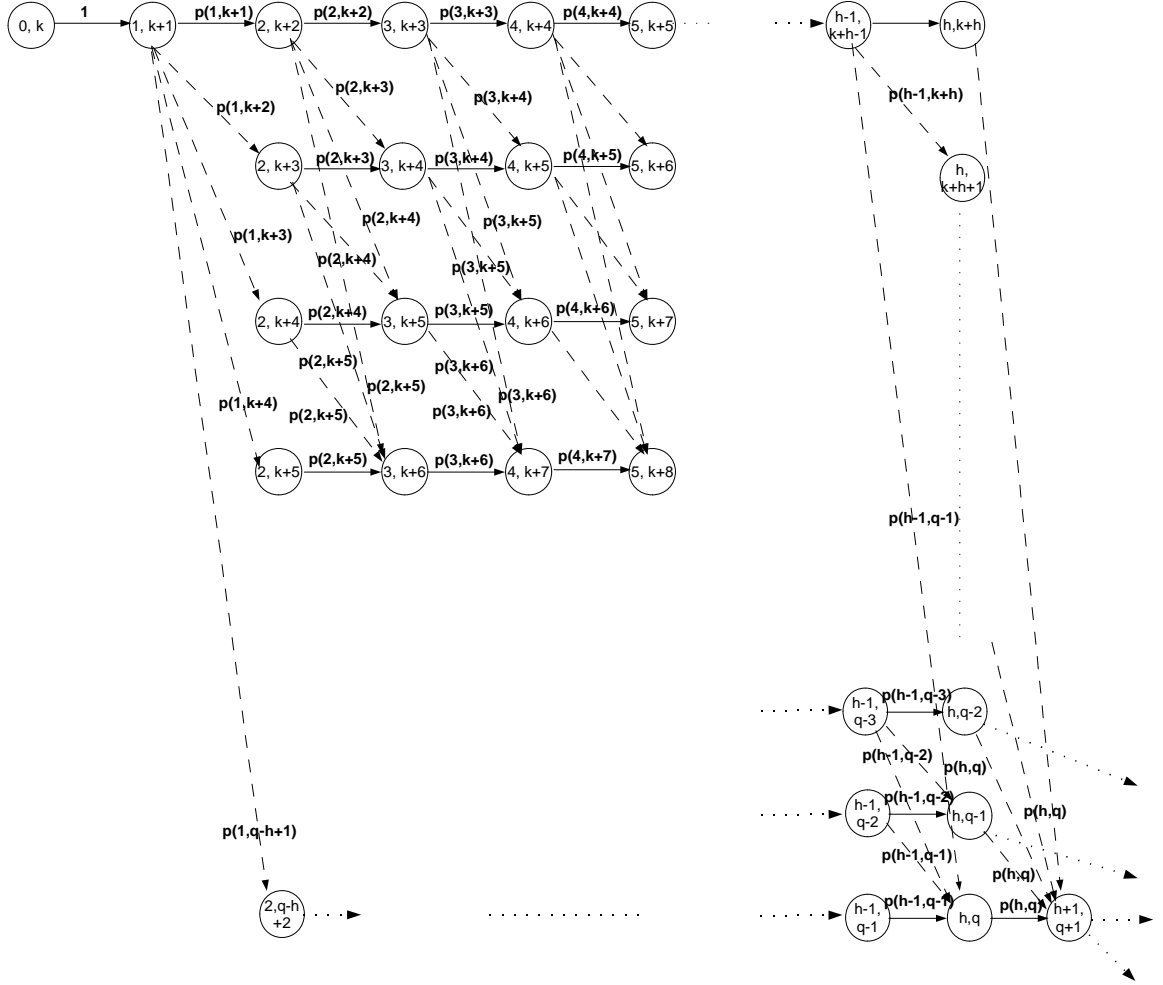


Figure 6.3: Markov Chain showing the propagation of a symbol x_i transmitted in the time slot k .

for $k = 1$. Further, the probability that the symbol propagates to other nodes in the network reduces drastically with an increase in the number of hops.

Considering that all nodes in the network get scheduled in n time slots, the author obtains the probability that a symbol x_i transmitted by the source i in the k th time slot propagates over h hops after time slot n . Limiting of the number of time slots to n is due to the primary motivation of measuring the propagation of information purely as a result of the minimum information transmitted. Typically, depending on the protocol specifications, nodes are likely to be scheduled multiple times so as to ensure successful delivery of all source symbols. However, as per the discussion in section 6.1, these would constitute additional overheads incurred. Hence, the propagation of

information when all nodes are scheduled exactly once is identified.

The propagation of x_i is modelled using a bidimensional discrete Markov Chain. A state $(h, q + 1)$ is denoted as the propagation state of the symbol x_i , indicating that it reaches hop h in the time slot q . $q + 1$ is used instead of q to denote that x_i has reached hop h before time slot $q + 1$. The probability that x_i reaches the state (h, q) is denoted by $b_i(h, q)$. The resulting Markov chain is shown in Fig. 6.3.

The probability that a node is scheduled to transmit in any time slot k is $p = \frac{1}{n}$, given uniform scheduling of the nodes. Given that node i transmits in the k th time slot, the symbol x_i is propagated to all nodes in the first hop as part of this transmission. Among the nodes in the first hop, x_i would only be propagated by the nodes which do not get scheduled in any of the preceding k slots. A subset of these nodes would get scheduled in the $(k + 1)$ -th time slot. Among the rest, a subset would again get scheduled in $(k + 2)$ and so on until n . The probability that a node in hop h' is scheduled in the time slot k' is denoted by $p(h', k')$ and can be obtained as,

$$p(h', k') = \frac{n - k' + 1}{n^2} \quad (6.1)$$

Although the expression in Eqn. (6.1) is independent of h' , h' is used to denote it for the convenience of explanation.

The probability of reaching a state $(2, k')$, $k' > k$ can be obtained as $b_i(2, k') = p(1, k') \cdot b_i(1, k + 1) = p(1, k') \cdot b_i(0, k)$ since all nodes in the first hop receive x_i in the k -th slot. However, for hop $h' > 2$, the probability of reaching the state (h', k') is dependent on the reception of x_i in the previous hop $(h' - 1)$. The earliest time slot in which a node in $(h' - 1)$ can receive x_i can be $(h' - 1) + (k - 1)$ which corresponds to the state $(h' - 1, h' - 1 + k)$ in Fig. 6.3. Also, for a node in $(h' - 1)$ to propagate x_i in $(k' - 1)$ (i.e. reaching state (h', k')), the latest it can receive x_i is in the time slot $(k' - 2)$ thereby reaching state $(h' - 1, h' - 1 + k' - 1)$. Thus, the probability of reaching the state (h', k') is the total probability of propagation from all the states $(h' - 1, h' - 1 + k)$ to $(h' - 1, h' - 1 + k' - 1)$.

The author now derives the probability that x_i reaches a state $(h + 1, q + 1)$, which

forms the core of the analysis. Based on the above discussion, this can be written as,

$$\begin{aligned}
 b_i(h+1, q+1) &= \sum_{k'=k+h}^q p(h, q) \cdot b_i(h, k') \\
 &= p(h, q) \sum_{k'=k+h}^q b_i(h, k')
 \end{aligned} \tag{6.2}$$

Note that regardless of the state (h, k') , the probability that x_i is propagated by a node in h in the time slot q is always $p(h, q)$. Examining the above expression closely and referring to Fig. 6.3, the author noticed it can be written as the sum of horizontal and angular chains, $b_i(h+1, q+1) = d_i(h+1, q+1) + c_i(h+1, q+1)$, where $d_i(h+1, q+1)$ and $c_i(h+1, q+1)$ are the summation over the horizontal and the angular chains respectively. This observation is used to split the derivation into two parts.

The summation of the set of horizontal chains is first obtained. Each horizontal chain culminates in one of the states (h, k') and corresponds to the sequence of transmissions in which any state (h', k'') , $2 \leq h' \leq h$, $(k' - h + 2) \leq k'' \leq k'$ can be obtained as $b'_i(h', k'') = p(h' - 1, k'' - 1) \cdot b'_i(h' - 1, k'' - 1)$. This implies that, after being propagated from the 2nd hop, in every subsequent hop, x_i is propagated by the set of nodes scheduled in the succeeding time slot. The notation b'_i is used to distinguish it from b_i indicating that it is only obtained from the horizontal links in the Markov chain. Thus, $d_i(h+1, q+1)$ can be expressed as,

$$d_i(h+1, q+1) = p(h, q) \sum_{k'=k+h}^q p(h' - 1, k' - 1) \cdot b'_i(h' - 1, k' - 1) \tag{6.3}$$

Each individual term inside the summation corresponds to one of the horizontal chains and can be expanded as,

$$\begin{aligned}
 &p(h' - 1, k' - 1) \cdot b'_i(h' - 1, k' - 1) \\
 &= p(h' - 1, k' - 1)p(h' - 2, k' - 2) \cdots p(1, k' - h + 1)b_i(0, k)
 \end{aligned} \tag{6.4}$$

Substituting Eqn. (6.1) in Eqn. (6.4) gives,

$$\begin{aligned}
 & p(h' - 1, k' - 1) \cdot b'_i(h' - 1, k' - 1) \\
 &= \frac{(n - k' + 2)}{n^2} \frac{(n - k' + 3)}{n^2} \dots \frac{(n - k' + h)}{n^2} b_i(0, k) \\
 &= \frac{b_i(0, k)}{n^{2(h-1)}} [(n - k' + 2)(n - k' + 3) \dots (n - k' + h)]
 \end{aligned} \tag{6.5}$$

Here, $b_i(0, k)$ is the probability that i transmits x_i in the k th slot, which is again obtained as $\frac{(n-k+1)}{n^2}$. Restricting the product term in Eqn. (6.5) to its dominating terms and reducing the resulting expression, gives

$$\begin{aligned}
 & p(h' - 1, k' - 1) \cdot b'_i(h' - 1, k' - 1) \\
 & \geq \frac{[(2n - h + h^2 - (h - 1)k')(n - k + 1)]}{n^{h+2}}
 \end{aligned} \tag{6.6}$$

Substituting Eqn. (6.6) in Eqn. (6.3),

$$\begin{aligned}
 & d_i(h + 1, q + 1) \\
 & \geq \frac{(n - q + 1)}{n^2} \sum_{k'=k+h}^q \frac{[(2n - h + h^2 - (h - 1)k')(n - k + 1)]}{n^{h+2}}
 \end{aligned} \tag{6.7}$$

which reduces to

$$d_i(h + 1, q + 1) \geq \frac{[2(q - (h + k - 1))(n - k + 1)]}{n^{h+2}} \tag{6.8}$$

Having obtained the expression for the summation of the horizontal chains, the focus is on the set of angular chains. Unlike the horizontal chains, the angular chains consist of the set of nodes that do not transmit immediately after receiving the symbol x_i . Among these, the author can further identify sets of chains characterized by the delay between reception of the symbol and propagating it. For instance, the state $b_i(h, q)$ can be reached by the chain represented as $p(h - 1, q - 1) \cdot b_i(h - 1, q - 2)$ corresponding to the set of nodes in the $(h - 1)$ th hop that receive the symbol in the

time slot $(q - 3)$ but get scheduled after remaining idle during the next time slot. The corresponding chain for the state $b_i(h, q - 1)$ is $p(h - 1, q - 2) \cdot b_i(h - 1, q - 3)$. Similar chains can be identified for all states $b_i(h, k + h + 1)$ to $b_i(h, q)$. While the above set of chains correspond to a delay of a single time slot, another set of chains can be identified for a delay of two time slots. For the state $b_i(h, q)$, this is given as $p(h - 1, q - 1) \cdot b_i(h - 1, q - 3)$. Similarly, different sets of chains are identified corresponding to the different values of the delay with the last set consisting of a single chain reaching the state $b_i(h, q)$ as $p(h - 1, q - 1) \cdot b_i(h - 1, k + h - 1)$.

The author first examines the set of chains corresponding to the delay of a single time slot. It is observed that each chain in the set can be expressed as a product of angular and horizontal chains terminating on the topmost horizontal chain. Evaluating this product of terms and summing over all the $(q - (h + k))$ chains in this set, and limiting to only the dominant terms, yields,

$$\begin{aligned} \sum_{k'=k+h+1}^q p(h - 1, k' - 1) \cdot b_i(h - 1, k' - 2) \\ \geq \frac{(q - (h + k))(n - k + 1)}{n^{h+1}} \end{aligned} \quad (6.9)$$

The next set has one less chain and the corresponding result of summation, having limited the expression to its dominant terms, is obtained as $\frac{(q-(h+k)-1)(n-k+1)}{n^{h+1}}$. As mentioned above, the last set consists of a single chain and thus, the summation of all the chains is obtained as,

$$\begin{aligned} \frac{(n - k + 1)}{n^{h+1}} \sum_{j=1}^{q-(h+k)} j \\ = \left(\frac{(n - k + 1)}{n^{h+1}} \right) \left(\frac{(q - (h + k))(q - (h + k) + 1)}{2} \right) \end{aligned}$$

As with Eqn. 6.2, the final expression for the state $b_i(h + 1, q + 1)$ would involve the

term $p(h, q)$. Thus,

$$\begin{aligned} & c_i(h+1, q+1) \\ & \geq \left(\frac{(n-k+1)(n-q+1)}{n^{h+3}} \right) \left(\frac{(q-(h+k))(q-(h+k)+1)}{2} \right) \end{aligned} \quad (6.10)$$

To conclude the analysis, the author needs to obtain the total probability that x_i propagates to a hop h once all nodes have been scheduled, in n time slots. The probability that x_i has propagated to hop $(h+1)$ after q time slots can be given as,

$$\begin{aligned} B_i(h+1, q+1) &= \sum_{q'=h+k}^{q+1} b_i(h+1, q') \\ &= \sum_{q'=h+k}^{q+1} d_i(h+1, q+1) + c_i(h+1, q+1) \end{aligned} \quad (6.11)$$

since the earliest time slot in which x_i can be propagated to hop h is $(h+k)$. Summing the horizontal chains first, results in,

$$\sum_{q'=h+k}^{q+1} d_i(h+1, q+1) \geq \frac{((q+1)-(h+k))^2(n-k+1)}{n^{h+1}} \quad (6.12)$$

The corresponding summation for the angular chains yields,

$$\sum_{q'=h+k}^{q+1} c_i(h+1, q+1) \geq \frac{((q+1)-(h+k))^3(n-k+1)}{6n^{h+2}} \quad (6.13)$$

Substituting Eqns. (6.12) and (6.10) in Eqn. (6.11) yields,

$$\begin{aligned} & B_i(h+1, q+1) \\ & \geq \frac{((q+1)-(h+k))^2(n-k+1)}{n^{h+1}} \left[1 + \frac{((q+1)-(h+k))}{6n} \right] \end{aligned} \quad (6.14)$$

The total probability once all nodes have been scheduled is obtained by substituting $q = n + 1$. Numerical results are shown in Fig. 6.4(a) for the different values of k .

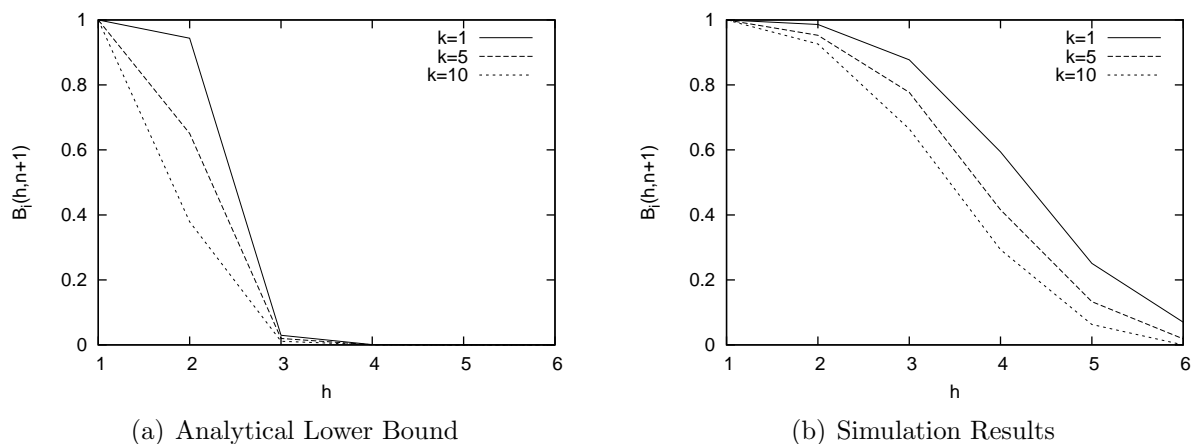


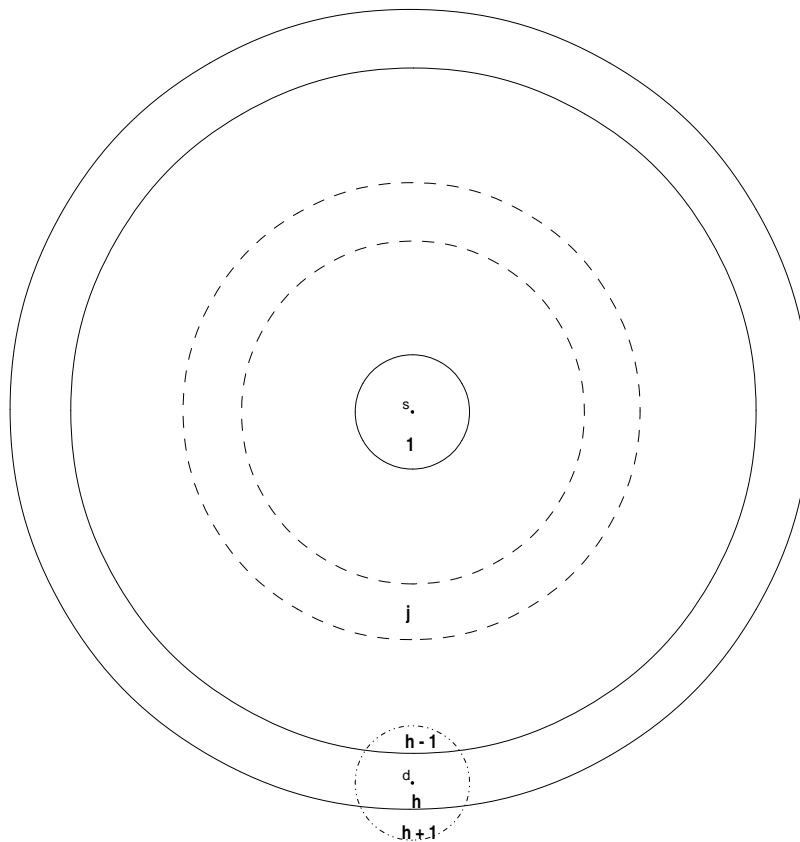
Figure 6.4: Illustration of the growth of the broadcast cache for different values of k .

The value of n is obtained using simulations for a network with nodes distributed randomly in a square region. The probability that a symbol propagates over multiple hops can be seen to have been reduced by about half for $k = 10$ compared to $k = 1$. While Fig. 6.4(a) shows the analytical lower bound, simulation results are shown in Fig. 6.4(b). Even though the symbol reception probabilities decrease similarly for successive hops, significantly higher probabilities of reception of source symbol are observed when compared to the analytical lower bound. This results from additional exchange of information among nodes in the same hop, which is not accounted in the current discussion.

6.3 Broadcast Cache as a measure of exploiting Wireless Broadcast Advantage

6.3.1 Feasibility Conditions for Exploiting Wireless Broadcast Advantage (WBA)

In this section, the author discusses how the results obtained in the previous section are effective for understanding the extent to which WBA is exploited in a multihop scenario. Given the set of information implicitly available at a node, the author ana-

Figure 6.5: Propagation of source symbol x_s to destination d

lyzes how it impacts the final performance. The author identifies the conditions that determine whether exploiting WBA using encoded transmissions is more beneficial for a flow performance when compared to traditional routing.

As mentioned earlier, consider that the network consists of N unicasts with each node having a packet to send to a randomly chosen destination. As received transmissions are randomly encoded combinations of source symbols, a destination node would need to acquire the required number of distinct packets containing the original source symbol. In the case of linear network coding, a node needs to have enough distinct packets to achieve a full rank. Consider that for a generic coding mechanism, a total of m distinct packets are required for a node to successfully decode a source symbol.

Consider the source symbol x_s for a unicast flow with source destination pair (s, d) separated by h hops, and the source scheduled to transmit in the k th time slot. This

is denoted by the tuple (x_s, s, d, h) . As a result of the minimum set of transmissions, the number of encoded combinations stored at d is the number of packets it receives from its first hop neighbourhood. Let \mathcal{N}_d^1 denote the set of first hop neighbours of d . The number of distinct packets stored in the cache at d is, therefore, $|\mathcal{N}_d^1|$. The probability that any of the packets include x_s is $B_s(h, n + 1)$. Denoting by $r_d^s(0)$ the number of packets stored at d that contain x_s , results in,

$$E[r_d^s(0)] = B_s(h, n + 1)|\mathcal{N}_d^1| \quad (6.15)$$

where 0 is used to indicate that the set of packets are stored in d 's own cache, i.e. the 0th hop. The decoding requirements at d would be satisfied if $E[r_d^s(0)] \geq m$, implying that it can successfully recover x_s from the set of stored packets. If this condition is not satisfied, however, the additional set of packets would need to be fetched by d from the other nodes in the network. As per the discussion in Section 6.1, the corresponding set of transmissions would constitute the protocol overheads. Furthermore, depending on the distance to the nodes that contain the packets stored in their cache, the network costs are compounded due to multihop transmissions. Thus, if the set of distinct packets can all be retrieved from the nodes located in hops $[1, l]$, the total number of transmissions required in the network would be

$$M_s(l) = \sum_{j=1}^l j|\mathcal{R}_j^s| \quad (6.16)$$

where \mathcal{R}_j^s is the set of distinct packets containing x_s stored in all nodes in the j th hop from d . Thus, allowing the flow (x_s, s, d, h) to exploit WBA by encoding it with other flows in the network is only beneficial for the flow performance if the following conditions are true

$$\begin{aligned} & \sum_{j=1}^l |\mathcal{R}_j^s| + r_d^s(0) \geq m \\ \text{s.t. } & M_s(l) < h \end{aligned} \quad (6.17)$$

since the minimum number of transmissions required for routing x_s from s to d is h .

The author discusses here how these conditions can be obtained if the set of additional transmissions are to be limited to the immediate neighbourhood of a node. As with the analysis, the author considers the propagation of a symbol between successive hops. For a node located in a hop j from the source s , the probability that x_s has propagated in n time slots is $B_s(j, n + 1)$. Thus, the total number of distinct packets in hop j containing x_s propagated from the preceding hop ($j - 1$) is $B_s(j, n + 1)|\mathcal{N}_s^{(j-1)}|$ where $\mathcal{N}_s^{(j-1)}$ is the set of nodes located ($j - 1$) hops away from s . Considering the destination d , it is observed that any node in its first hop neighbourhood \mathcal{N}_d^1 would be located in a hop $i \in [h - 1, h + 1]$ from the source (Refer to Fig. 6.5). Among these, node d is likely to benefit most from nodes located in the region of hop ($h - 1$). This is because it is likely to have already received a majority of the packets stored by neighbours located in hop h from s . As the packets stored in ($h + 1$) are also propagated from h , these are also likely to have been received by d . Focusing on the region lying in the hop h from s , the corresponding area can be obtained from the area of intersection of two circles

$$\begin{aligned}
 & A(x, r, R) \\
 &= r^2 \cos^{-1}\left(\frac{x^2 + r^2 - R^2}{2xr}\right) + R^2 \cos^{-1}\left(\frac{x^2 + R^2 - r^2}{2xR}\right) \\
 & - \frac{1}{2} \sqrt{(-x + r + R)(x + r - R)(x - r + R)(x + r + R)} \quad (6.18)
 \end{aligned}$$

where x is distance between the centres of the two circles. The corresponding area of the region in \mathcal{N}_d^1 located ($h - 1$) hops away from s is given as $A(x, h - 1, 1)$ where x is the distance between s and d and the transmission range is normalized to 1. Thus, among the set of distinct packets in hop ($h - 1$) containing x_s , the fraction accessible to d from its first hop neighbourhood is $\frac{A(x, h-1, 1)}{(2h-3)\pi}$ where $(2h - 3)\pi$ is the area of the region in the ($h - 1$)th hop. From the earlier discussion, the total number of distinct packets in the hop ($h - 1$) is given as $B_s(h - 1, n + 1)|\mathcal{N}_s^{(h-2)}|$. Given a node density ρ , $|\mathcal{N}_s^{(h-2)}| = (2h - 5)\pi\rho$. Denoting by $r_d^s(1)$ the number of distinct packets that d

can obtain from its first hop neighbourhood, results in,

$$r_d^s(1) \geq \frac{A(x, h-1, 1)}{(2h-3)\pi} B_s(h-1, n+1) |\mathcal{N}_s^{(h-2)}| \quad (6.19)$$

Thereafter, the optimal conditions for the flow (x_s, s, d, h) can be obtained using Eqns. (6.15), (6.17) and (6.19) as,

$$\begin{aligned} r_d^s(1) + r_d^s(0) &\geq m \\ \text{s.t. } r_d^s(1) &< h \end{aligned} \quad (6.20)$$

The above discussion is focused on the tradeoffs involved in the number of transmissions required for successful packet delivery. This corresponds directly with the flow performance as the throughput is inversely proportional to the number of transmissions involved. Further, the delay performance of a flow improves with reduction in the number of transmissions. However, the precise throughput and delay performance depend on individual scenarios and protocol specifications. For instance, any additional transmission is likely to incur contention overheads. Comparison of traditional routing with that of exploiting WBA would involve analysis of the contention overheads incurred by the protocol specifications in both scenarios. Nevertheless, a detailed assessment of individual parameters is not within the scope of the current discussion.

6.3.2 Design Considerations for Distributed Algorithm Design

The results in the previous section show that the effectiveness of the wireless broadcast advantage for a flow depends on the latter's characteristics. Specifically, the scheduling of the source node and the distance between the source and the destination determine whether a flow should be enoded along with the other flows in the network or should be routed as an individual packet. Although algorithm design

based on the analysis would depend on individual encoding schemes, certain practical considerations are discussed here.

An important aspect of the analysis is that the author bases it on the number of 2-independent sets in the network, which is impossible to estimate from the perspective of an individual node. However, node decision making from the perspective of algorithm design can be based on the 2-hop neighbourhood size of a node. This is because, for any node in the network, none of its neighbours belong to the same 2-independent set as itself. Thus, the 2-hop neighbourhood size is never greater than the number of 2-independent sets in the network.

Another important consideration from the point of view of algorithm design is how the set of additional symbols are retrieved. The conditions outlined in Eqn. (6.17) are based on the assumption that the stored packets in each hop are individually obtained. However, depending on the coding mechanism used, the packets could be encoded again depending on the flow requirements of the neighbourhood.

6.4 Future Work

As mentioned earlier, the analysis in this chapter makes use of certain ideal conditions. The author proposes relaxing these conditions as part of the future work.

Ideal channel conditions such as error free transmissions are assumed in the current analysis. This is highly unlikely to be true in realistic scenarios. Instead of the unit disk graph (UDG) model considered in this chapter, transmission irregularities can be modeled using the quasi-unit disk graph (QUDG) model [100]. Furthermore, the assumption of uniform and fair scheduling can be replaced by a realistic scheduling mechanism such as the IEEE 802.11 DCF.

The analysis here assumes that nodes store all overheard information. This implies that each node needs to have a minimum buffer size equal to the number of neighbours. However, it would need additional buffer to receive and store additional packets required for decoding a source symbol. Thus, the feasibility conditions need to be re-evaluated from the point of view of the memory status at each node.

6.5 Conclusion

In this chapter, the author obtains a measure of the broadcast advantage in a multihop wireless network. The author considers a setup in which nodes transmit encoded combinations of all neighbourhood transmissions, resulting in propagation of information over multiple hops. The author shows how the results can be useful for determining whether exploiting WBA can result in improved performance for a flow depending on the flow characteristics.

Chapter 7

Conclusion & Future Work

The author concludes the dissertation by summarizing the key contributions and proposing directions for future research work.

7.1 Summary of Contributions

This dissertation presented novel ideas on the design and analysis of algorithms that optimize wireless network performance by minimizing the overhead costs. The author's approach consisted of identifying problems that are crucial to the overall network performance. The major contributions addressing each of these issues are summarized as follows:

7.1.1 Differentiated Channel Access

Differentiated channel access to ensure service guarantees for real-time traffic is addressed in Chapter 3. A contention-tone (CT) based service differentiation scheme is proposed. Key features of the proposed solutions are outlined as follows:

- Dual degrees of freedom to the design of CTP allows flexibility to design requirements.
- Optimal choice of parameters to achieve bandwidth and device differentiation.

- Proposed scheme is shown to provide effective bandwidth differentiation even at high node densities.
- Access point bottleneck problem is solved to balance uplink and downlink performance for two-way real-time traffic.

7.1.2 Stateless Network-wide Broadcast

In Chapter 4 of this dissertation, the author studies the design of network-wide broadcasting algorithms in which nodes decide on their forwarding behavior based on information obtained by exploiting the wireless broadcast advantage (WBA). These are termed as stateless algorithms as no prerequisite information is needed at nodes, thereby minimizing the overhead costs. The author's primary contributions are:

- A unified analytical model that identifies the performance benefits achievable from different phases of broadcasting in ideal channel conditions.
- Insights on design of stateless algorithms in terms of impact of node density on parameter choice and the permissible limits of additional overhead costs.
- Extension of the above model to take care of dynamic channel conditions and insights on parameter choice with respect to channel conditions.
- Analysis of network-wide broadcast in a multi-rate scenario. The author identifies broadcast effectiveness tradeoffs for transmission at the different data rates.
- Design and evaluation of a multi-rate broadcasting algorithm based on the relation between broadcast effectiveness at different data rates.

7.1.3 Self-organization of Wireless Networks as Small Worlds

A self-organization framework for wireless ad hoc networks to realize small world behavior is proposed in Chapter 5. The author investigates the use of directional beamforming antennas for creating long range shortcuts. The author's primary contributions can be summarized as:

- Simulation based analysis of use of randomized directional beams for small world creation. Achievable performance benefits and important design challenges are identified.
- Proposal of new centrality measure for wireless ad hoc networks called Wireless Flow Betweenness (WFB). WFB can be computed at nodes in a purely distributed manner by exploiting the wireless broadcast advantage (WBA). It is shown to give a close approximation to the centrally computed Flow Betweenness Centrality (FBC).
- A distributed algorithm using WFB measures computed at nodes which is shown to give desired benefits for small world creation in wireless ad hoc networks.

7.1.4 Wireless Broadcasting Advantage (WBA)

Motivated by the benefits of exploiting WBA, the author proposes an analytical framework to measure WBA in Chapter 6. The author studies the propagation of opportunistically received information over multiple hops. Key aspects of the research are:

- Treatment of WBA in terms of a distributed cache, termed *broadcast cache*, which is built up across the network as nodes store and forward implicitly received information.
- Analysis of the propagation of information purely as a result of WBA, in terms of growth of the *broadcast cache*.
- Feasibility conditions that determine whether WBA can be exploited by a traffic flow depending on the flow characteristics.

7.2 Future Research Directions

Here, the author proposes future research directions which can be pursued based on the contributions in this dissertation.

7.2.1 Contention tone Protocol with Service Differentiation

Future directions for research on contention tone protocol are discussed in detail in Section 3.4 and are summarized as follows:

- In addition to prioritized channel access, admission control constitutes an important aspect of providing QoS guarantees at the MAC layer. By determining the behavior of the contention tone protocol on the basis of other QoS parameters specified in the TSPEC, a wholesome solution that takes into consideration all aspects of QoS design can be achieved.
- To make the proposed solution flexible to future network deployments, its performance needs to be investigated in the context of increasing data rates.

7.2.2 Stateless Network Wide Broadcast

The author concludes in Chapter 4 that the design parameters for broadcast algorithms need to be dependant on the node density. Furthermore, forwarding behavior at nodes needs to adapt to channel conditions which may be dynamic. Thus, nodes need to estimate their neighborhood conditions for optimized decision making. Recent research has shown that node density can be estimated based on channel contention [129], thereby incurring zero transmission overheads. However, the proposed approach in [129] requires a high number of time slots for accurate estimation and a more proactive approach may be required in mobile scenarios and for probing instantaneous channel conditions. In this case, the amount of overheads would also determine the parameter values.

Stateless multi-rate broadcasting protocols can be designed based on the algorithms proposed in Section 4.2 and the insights obtained on stateless broadcasting in general in Section 4.1. Furthermore, scheduling is an important aspect of minimum latency broadcasting (MLB). Based on the author's proposed algorithms, effectiveness of prioritized channel access among nodes contending to act as forwarding nodes may be investigated.

7.2.3 Self-Organization in Wireless Networks

While the author explores the self-organization of wireless networks as small worlds, an obvious next step is to explore the routing benefits achievable in such a setup. The navigability of small worlds using only local information was investigated in the seminal paper by Kleinberg [115]. The overheads incurred by routing in ad hoc networks have been studied in the literature [4, 5]. An interesting research direction is to explore whether these overheads can be alleviated if nodes are aware of the location of shortcut links.

Although the proposed WFB measure has been used in this dissertation for identifying the optimal set of nodes to create shortcuts, it is likely to be useful for addressing other issues in wireless ad hoc networks. This is because, as shown in Section 5.2.2, the WFB values give a good measure of the structural importance of nodes in the network irrespective of the final application. The structural importance of nodes in the network has been shown to be useful for various applications such as caching in sensor networks [128] and routing in delay tolerant networks (DTNs) [130, 131, 132]. Given its completely distributed nature, WFB presents an attractive choice for such protocol designs.

7.2.4 Wireless Broadcast Advantage (WBA)

The analysis in Chapter 6 gives a measure of WBA under ideal channel conditions. A more accurate measure can be obtained by considering dynamic channel conditions which impact implicit reception of neighborhood transmissions.

From the point of view of algorithm design, the analysis can be extended to consider the limited buffer at the individual nodes which can influence the creation of the *broadcast cache* itself. Thereafter, algorithms that exploit WBA can be designed for nodes to store information relative to the network requirements.

A practical extension is to use the feasibility conditions of exploiting WBA for protocol design. Exploring algorithm design that adapts dynamically to network conditions and flow characteristics is an interesting research direction.

Author's Publications

Journals

- A. Banerjee, R. Agarwal, V. Gauthier, C. K. Yeo, H. Afifi, B. S. Lee, “A Self-Organization Framework for Wireless Ad Hoc Networks as Small Worlds,” *IEEE Transactions on Vehicular Technology*, accepted with minor revisions.
- R. Agarwal, A. Banerjee, V. Gauthier, M. Becker, C. K. Yeo, and B. S. Lee, “Achieving Small World Properties using Bio-Inspired Techniques in Wireless Networks,” *The Computer Journal*, accepted.
- A. Banerjee, C.H. Foh, C. K. Yeo and B. S. Lee, “Performance Improvements for Network-Wide Broadcast with Instantaneous Network Information,” *Elsevier Journal of Network and Computer Applications (JNCA)*, vol. 35, no. 3, pp. 1162-1174, May 2012.
- A. Banerjee, J.W. Tantra, C.H. Foh, C.K. Yeo and B.S. Lee, “A Service/Device Differentiation Scheme for Contention-Tone Based Wireless LAN Protocol,” *IEEE Transactions on Vehicular Technology*, vol. 59, no. 8, pp. 3872-3885, Oct. 2010.

Conferences

- A. Banerjee, R. Agarwal, V. Gauthier, C. K. Yeo, H. Afifi and B. S. Lee, “Self-Organizing of Wireless Ad Hoc Network as a Small World using long

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